by Joe Cooper, carm_popcomm@hotmail.com

Digital Signal Processing— The Theory Behind Today's Digital Communication Technology

I had originally intended for this month's column to be a continued look at how to use Microsoft's Visual BASIC Express software development program to help you create a CAT (computer-assisted tuning) program for Ten-Tec's RX-320 "black box" communications receiver. However, I was getting feedback that I was moving ahead of the learning curve for some people, and they needed a break.

This is understandable as a number of readers had just downloaded their new Visual BASIC programs and were still working their way through the basics of how to use the application. So people were starting to fall behind as I began our look at more advanced topics, such as user interfaces (UIs) for the CAT program. Plus, there are many of you out there who aren't going to become computer programmers, but still want to learn more about the technology behind software-defined radios (SDRs).

For those who may have tuned in late, Microsoft is offering Visual BASIC Express, along with other software development language applications, on its website http://msdn.microsoft.com/vstudio/express/default.aspx. This particular version of Visual BASIC is unique in that it's completely free to download and is designed to help you to learn computer programming at a hobbyist level. That goal is facilitated through an extensive free training program.

So let's take a short break from the computer programming series; we'll come back to it in about four issues to give everyone a chance to get caught up. In the meantime, since I've already outlined the basics of what needs to be done in my last three columns, if anyone gets ahead of the game and puts together a working CAT program, please tell me about it. I'll be happy to share your story with the readers of this column. It doesn't have to be fancy, just work reasonably well.

Looking At DSP Technology

Over the next four columns I want to revisit digital signal processing, (DSP), which is a not only a very important component of SDR, it's also used extensively in any type of data communication today, whether voice, picture, or sound based.

SDR technology is totally dependent upon DSP technology in order to work. SDRs use DSP technology to produce the phased digital information needed for the software-based I/Q algorithms that demodulate the intelligence, such as voice or music information, contained in a digital signal. The front end of any SDR radio is based upon "ordinary" DSP technology, so it's important to know how it works. DSP technology is not that difficult to understand—it's simply where an analog signal is converted into a digital form and then manipulated using high-speed mathematical processing.

If you have a cell phone, a CD player, a handheld video game player, or even many modern superheterodyne radios, you're

already using DSP technology. Why is DSP so important in today's modern communication devices? Simply because DSP works to dramatically improve the sensitivity of a receiving unit in ways that could never be achieved with old analog circuits, whether they were vacuum tube or solid state devices. The truth is, all communications circuits contain some noise, whether the signals are analog or digital and regardless of the type of information conveyed. It's a problem with any communications technology and there's an ongoing struggle to improve the signal-to-noise (S/N) ratio of any communications systems.

Back in the old days of spark transmitters, the first method of optimizing S/N ratio included increasing the transmitted signal power up to a million watts. However, given the harmonic interference, such powerful signals made that approach impractical. So since the earliest days of radio the focus has been on improving the *receiving* side of the communications circuit.

DSP technology, while theoretically complex, is actually very simple from the hardware standpoint. For example, in processing an incoming analog signal, say a shortwave broadcast, the signal is first converted to digital form by an analog-to-digital converter (ADC). The resulting digital signal has two or more levels, which ideally are always predictable, exact voltages or currents. However, because the incoming signal contains noise, the levels are not always at the standard values. A DSP circuit works by adjusting these levels so they're at the correct values, which practically eliminates any noise. The digital signal is then converted back to analog form via a digital-to-analog converter (DAC), which allows it to be filtered or amplified in a conventional manner.

However, if a received signal is digital, say computer data, the ADC and DAC are not necessary. You can instead use the

DSP circuit to act directly on the incoming digital signal, eliminating irregularities caused by noise, and thereby minimizing the number of errors that may be contained in that signal.

A good DSP circuit will be superior to any conventional analog filter, but there are limits as to what it can do. Needless to say, if the noise is so strong that all traces of the signal's "intelligence" are obliterated, the DSP circuit will not be able to readjust (or process) that intelligence to a usable level. As a result all that can be passed through the DSP circuit is the original noise, so the bottom line is that you must have some signal in a digital signal in order to filter it properly using DSP.

The Beginnings Of Digital Technology In Telegraphy

What's really interesting about DSP is that while it's generally associated with today's modern computers, it's actually a much older technology. Strange as it may seem, the origins of DSP are traceable directly to theoretical work done in the 1920s on how to improve telegraphic signals.

Now you might think this a bit unusual, but the inspiration for taking a closer look DSP technology for this series of columns came when I read that Western Union had quietly ended telegraph service on January 27, 2006. Although this news item might not appear to have any direct connection to the topic of DSP, the truth is that if telegraphy had never been invented, then neither would DSP technology—the two technologies are that closely linked.

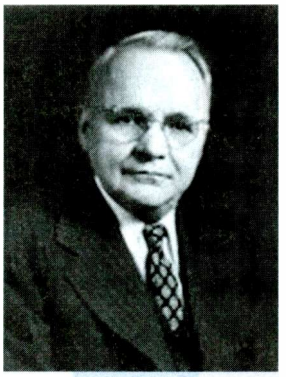
Telegraphy was originally developed in France in 1792 by Claude Chappe, and its introduction led to the establishment of countrywide telegraphy stations by the early 1800s. A true telegraphy system (based on the use of semaphores), it was a visual approach to communications, and as such was limited to daytime use, and even then could only be used in good weather. Alternative methods of telegraphic communication were, therefore, investigated in a number of countries.

Some early methods of electrical transmission of information were attempted in various forms in Europe. Baron Pavel Lvovitch Schilling, a Russian diplomat in Germany, developed the world's first practical electromagnetic system of telegraphy in 1832. But since the wires used to transmit telegraph signals could be cut, European military leaders saw his system as strategically vulnerable, so little progress was made in its development.

A year before Schilling developed his system, however, U.S. scientist Joseph Henry successfully demonstrated his own unique electrical telegraph system. In Henry's approach, electric current traveled down a wire, over even great distances, and then magnetically activated a striker, or plunger, creating a noise. Five years later, Samuel Morse would successfully exploit Henry's invention by creating a commercially viable telegraph network in the United States.

The differences between the Morse system and both Schilling's and Henry's were significant. For instance, Schilling used a compass needle that, as it turned through a circle of 360 degrees, would point at a letter of the alphabet, stopping on one and then moving on to the next letter of a word. Henry's system was no different from a simple doorbell: all it could do was ring a bell at a distance.

Perhaps Morse's greatest contribution was the code that bears his name. It cleverly exploited the human brain's amazing ability to distinguish patterns through the analysis of differences in groups of things it encounters. Normally such groups can be comprised of the sounds of a musical score, visual information such as letters in an alphabet, or even abstract ideas such as words in a crossword puzzle or figures in a mathematical formula. Morse's code was made up of groups of dots, dashes and spaces, which are simply created by the different lengths of time in which a current passes through a magnetic circuit.



Harry Nyquist (1889–1976) was born in Nilsby, Sweden, prior to his family's immigration to the United States. After obtaining a Ph.D. from Yale in 1917, he went to work for AT&T where he developed his now famous theory of digital sampling in 1928, which laid the foundation for today's DSP technology. In 1934, he went to work for Bell Laboratories, where he stayed until his retirement in 1954. (Photo courtesy IEEE)

The human brain, when properly trained, is smart enough to measure the minute differences in time lengths between the three elements in the original Morse code (later two elements, dots and dashes, would be used in the International Code). That same training allowed the brain to associate those different groups of elements into numbers or letters of the alphabet.

But Morse also exploited Henry's contribution to telegraphy by using a plunger to place pencil marks on a moving piece of paper, which would be "read" by someone familiar with the code it "spelled" out. Interestingly enough, Henry's legitimate invention of the magnetic plunger would be used in a lawsuit against Morse's patents, which ended with Morse losing his patents to the telegraph in the 1840s. This decision was overturned in 1853, however, when the U.S. Supreme Court ruled in Morse's favor out of respect for the way society had benefited from his development and application of the technology, which Henry simply had not done.

Regardless of its early controversies, telegraphy was "the original digital" in that "intelligence" was sent as individual bits of information; today those bits take the form of being in a state of either "on" or "off."

Telegraphy Gives Place To DSP

By the early 1920s data communication was becoming increasingly complicated, particularly with the advancements being made in radio technology. As a result, landlines were no longer carrying only the dots and dashes of Morse code, but also telephone and teletype signals. Indeed telegraph companies were even transmitting code using radio waves over their wires to increase their transmission bandwidth. However, this approach introduced the old adversary of any communication method: noise.

About this time, Harry Nyquist, a scientist employed by AT&T, undertook a series of studies on how to use digital sampling techniques to reduce or eliminate noise in telegraph signals. His work culminated in 1927 with the discovery that an analog signal should be sampled at regular intervals over time and at twice the frequency of the signal's bandwidth in order to be converted into an adequate representation of the signal in digital form. The results of this research appeared in a paper entitled "Certain topics in telegraph transmission theorym," which

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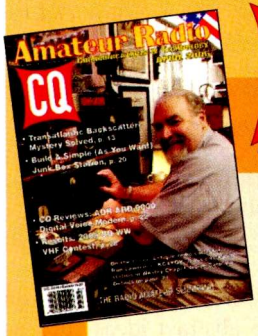

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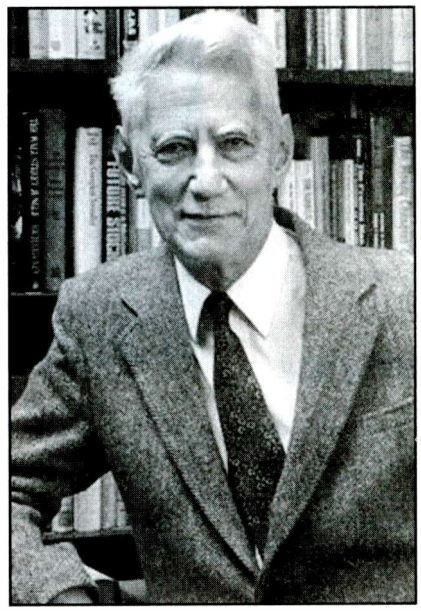
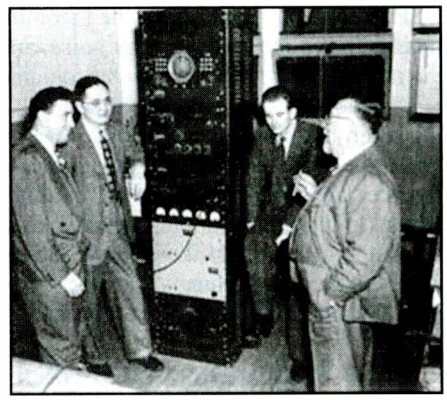


Claude Shannon (1916–2001) graduated from the University of Michigan in 1936 with bachelor's degrees in mathematics and electrical engineering. In 1948 he published his landmark piece, "A Mathematical Theory of Communication," and single-handedly started today's modern communication industry. (Photo courtesy Bell Laboratories/Lucent Technology)



Thanks to the ideas, influence, and theories of Harry Nyquist and Claude Shannon, post-World War II experiments at the Research in Electronics facility at MIT focused on DSP technology. This photo shows an early DSP experiment being conducted under the supervision of Professor Norbert Wiener, who developed the discipline of Cybernetics, the study of communication and control through feedback. (Photo courtesy RIE/MIT)

was published in 1928 and is available for download at www.comelec.enst.fr/~rodrigez/CNTI/images/nyquist.pdf.

Also around the same time a young man named Claude Shannon was working as a messenger for Western Union in his home town of Gaylord, Michigan, saving up to enter university for a degree in electrical engineering. After entering the University of Michigan, Shannon incorporated Nyquist's ideas on digital sampling to develop the basic concepts that have lead to today digital computers, CD recordings, and data compression. The early work of both Nyquist and Shannon was used by a number of communications companies, including Western Union, to improve their relaying of information. But Shannon's impact did not end simply with his contribution to digital circuitry; he also furthered early DSP research and design through his association with MIT and Bell Laboratories.

It was at MIT during the 1940s and '50s that Nyquist's and Shannon's work was applied to the creation of DSP in the first generation electronic digital computers. These advances lead to the development of the first of practical software-based DSP using the (then) high speed TX-2 computer. The success of this project laid the foundation for digital recording, signal processing, and data compression, culminating in today's digital revolution.

Applied Technology

It's interesting to note how that new digital revolution impacted one of the pioneers of the digital age, Western Union. Western Union used the ideas of Nyquist and Shannon to improve their rate of digital transmission and circuit switching.

By the 1940s telegraph keys had pretty much been retired for teleprinters, and faster code systems, such as Baudot and ASCII, came into use. These advancements in telegram technology evolved into a computerized system of digital transmission and switching that was to become the foundation for the military's Automatic Digital Network (AUTODIN) used during the 1960s.

That digital communications system was very slow by today's standards, running at 2400 baud, but it was reliable and reigned supreme until 1982, when it was replaced by ARPANET. ARPANET, as you may know, would become the foundation for the early Internet of the 1990s.

Another technology Western Union developed in conjunction with AUTODIN was one of the first true digital commercial communication systems. Called EasyLink, this integrated service used a modem to connect with a computer terminal over a telephone line that enabled the user to communicate by e-mail, fax, or telegram.

During the early 1970s the DSP program at Research in Electronics at MIT had use of the (then) powerful TX-2 mainframe computer. It was here in 1972 that the first software-based DSP application was created. This became the foundation for our current DSP technology and software-defined radios. (Photo courtesy RIE/MIT)

Ironically, it was the success of AUTODIN (which led to the creation of the Internet) and EasyLink (which led to today's e-mail systems) that eventually caused the demise of the Western Union telegraph. By 2005 only 20,000 telegrams a year were being sent. So on January 27, 2006, after 155 years of continuous service, Western Union ended its telegraph service. That final telegram sent by Western Union was not just an important milestone in the history of telegraphy, but in computer history as well. Truly, "what hath man wrought?"

Next Month

I'll continue this exploration of DSP technology, looking at how analog signals are sampled using digital sampling devices, then processed using DSP software before being turned back into post-processed analog signals. I'll introduce you to some interesting DSP software and hardware that can be used to enhance your current radio monitoring practices.

Do not forget to e-mail me with any questions at carm_popcomm@hotmail.com. As mentioned before, I cannot answer general questions on computers, but will be more than happy to help you with any issues raised in the columns.

I've heard that this summer some people in the weather business are predicting up to nine significant hurricanes, which is disquieting given that there are still many people still feeling the effects of past natural disasters like Hurricane Katrina. I would like to suggest that rather than waiting for a disaster to occur before contributing, you might send a donation now to the American Red Cross (www.redcross.org/donate/donate.html) to help your fellow Americans in this time of trouble. There are many good (and ethical) organizations that you can contribute to so please use them if you wish, but do not give into "charity fatigue."

If you have a job, a family around you, and are living in a stable neighborhood, then show your thanks for that wonderful luck by sharing of it someone less fortunate, and do so regularly.

Let us also not forget our troops overseas who continue to need our support, particularly as tensions in the Middle East continue to rise. Please refer to the U.S. Department of Defense's official webpage, "Defend America." There is a section found at www.defendamerica.mil/support_troops.html with an amazingly wide range of practical and useful ways that you can directly help.

Again, if you are fortunate enough to live in the United States of America, have a safe and secure home, a paying job and your loved ones around you when so many don't, please remember to give thanks for your personal blessings by passing on that blessing to others through regular acts of selfless sharing. See you again next month! ■

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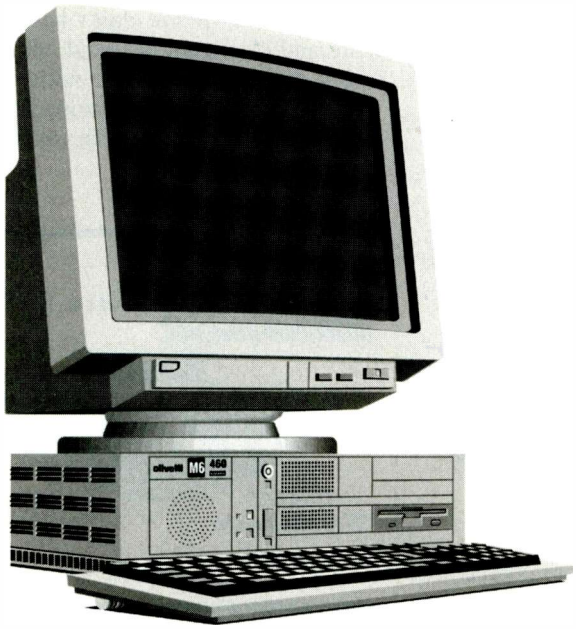
Digital Signal Processing, Part II—Learning The Theory Behind Today's Digital Communication Technology



No matter what type of modern communication technology you use today, whether cell phones, television, radio, PCs or CD players, all are becoming increasingly dependent upon digital signal processing (DSP). The same goes for software-defined radios (SDRs). All SDRs use DSP to produce the phased (I/Q) digital information characteristic of that technology. That phased signal is used to demodulate the intelligence contained in a digitized radio signal to turn it into usable information, such as voice or music.

I've touched on this topic before, for instance in the column on the set-up and operation of PC sound cards back 2002, as well as in 2003 when I outlined how to use those PC sound cards to digitally record audio signals. Even in the short time since I wrote those columns, there have been many improvements and refinements in the application of DSP, particularly in regard to radio-frequency applications.

DSP technology is itself not new at all. You can argue that it had its origins in the earliest forms of "the original digital": telegraphy. Right from introduction of telegraphy in the late 1700s, there were many attempts to improve the quality of the signal by processing it in different ways. The placement of telegraph signals on wires via electricity was the starting point of our current electronic-based DSP technology.

The Modern Era

Modern digital recording techniques had their origins in theoretical work undertaken in the 1920s to improve the reception of telegraph signals at distant locations. The most famous early contribution was that of AT&T scientist Harry Nyquist. His research into improvements in telegraphy led to a very important discovery in digital recording: an analog signal should be sampled at regular intervals over time and at twice the frequency of the signal's bandwidth in order to be converted into an adequate representation of that analog signal in digital form. That discovery was then used by Claude Shannon in the 1940s to develop the basic concepts that have led to today's digital computers, CD recordings, and data compression.

What kept people from actually applying DSP to mainstream electronic technology earlier than the 1970s was the massive size of the required analog devices, such as vacuum tubes. An example of how unwieldy this could be was the famous SIGSALY, one of the earliest practical applications of DSP technology. This digitally encrypted telephone system was developed by a team of Bell engineers led by A.B. Clark and used in World War II.

When it was first officially put into service on July 15, 1943, there were two terminals (one in Washington and one in London). Each was made up of 40 racks that contained a total of 384 model 2051 Gas Tetrode Thyratron vacuum tubes. The total weight of the equipment was approximately 55 tons, not including the air conditioning system required to keep it cool!

Each terminal was so complicated to operate and difficult to maintain that the 805th Signal Service Company was formed specifically to support SIGSALY. To give you an idea of the manpower needed to keep the terminals working properly, the 805th was made up of 81 officers and 275 enlisted men, divided into 12 detachments of five officers and 10 men to cover the 12 terminals that would eventually be in operation.

Because of the extreme importance of the SIGSALY project, the men of the 805th had the highest qualifications of any technical group in the signal corps. Each detachment was expected to keep each of the 384 vacuum tubes in each terminal working 100 percent of the time when the terminals were in operation. To achieve this, the detachments were expected to be available 24 hours a day, giving SIGSALY a service cycle of 16 hours of maintenance for every eight hours of operation. That cycle included testing all 384 vacuum tubes in the terminals every day and keeping 384 tube sockets in working order despite wear and tear.

However arduous the maintenance involved, this method of digital communication allowed military commanders in the United States to communicate with their British counterparts in a spontaneous way through telephone conversations. More importantly, this more natural way of communicating eliminated the time wasted in having written messages encrypted, transmitted, and then decrypted. In addition, the actual intelligence used in the SIGSALY transmissions was completely unde-

In 1943 two SIGSALY terminals were used to ensure 100 percent secure voice communication between military and political leaders in Washington and London. This system was the first successful use of DSP technology, comprising analog-to-digital sampling, signal processing, and data compression. It took a team of between 10 and 13 men to run SIGSALY and required 15 minutes just to start the equipment and synchronize the two terminals. (Photo courtesy National Security Agency)

tectable and undecipherable by the Germans and Japanese because they had no comparable digital technology to intercept the signals.

The transmission method was found sufficiently secure for President Franklin Roosevelt and Prime Minister Winston Churchill to use it to talk directly with each other by telephone, unlike the other wartime leaders who needed their messages encrypted by hand. In fact, SIGSALY technology was so effective as a means of encrypted voice communications that its basic methodology remained classified as a military secret until 1976, and it was not fully revealed to the public until 1983!

As you can see, a significant amount of research and development has gone into today's DSP technology. So in appreciation of the many people who were instrumental in its advancement, this month's column will take a closer look at why DSP technology has become so central to today's radio technology and how it operates at a theoretical level. By understanding how this technology works, you'll be able to make better use of its benefits in your radio monitoring. Let's begin by looking at why DSP technology is used in the first place.

What Is A Signal?

A signal is any disturbance that conveys an intelligent message. That can range from clapping your hands together to get someone's attention to communicating with a space probe at the edge of the solar system using radio signals. If you can't get someone's attention and have him or her react properly to the sound of your clapping hands, or get a space probe to respond to a radio signal that has traveled billions of miles, your signal has failed to convey intelligence. That failure could be because the signal was too weak to be heard by either the human ear or picked up by a spacecraft's antenna array. But maybe not.

"...The point is that no matter what kind of signal is being sent, its ability to convey intelligence always depends on its ability to overcome noise."

For instance, if you try to attract someone's attention in New York's Time's Square by clapping your hands, you'll be competing against other noises, like traffic, making it next to impossible for someone to recognize your signal. The same goes for our space probe's antenna; it may also be picking up cosmic noise, background radiation, and a host of other natural signal sources that create a kind of interstellar Time's Square. The point is that no matter what kind of signal is being sent, its ability to convey intelligence always depends on its ability to overcome noise. Obviously someone clapping his hands will increase his chances of success when there is little or no competing noise. Same with the space probe.

Noise in transmissions also affected early forms of visual telegraphy, where "noise" could take the form of rain, fog, smoke, or even darkness. Anything that could decrease that noise, including switching the transmission of intelligence from visual methods to an electrically based system over wires, would essentially be some type of signal processing.

The issue of signal versus noise is so central to any type of communication (again, this can be sound from a stereo, cell phone, TV, or radio) that the overall efficiency of a device is expressed by the signal-to-noise ratio it's capable of delivering. In fact, overcoming noise (defined as the background "hiss" created by vacuum tub technology) was such a significant part of the SIGSALY technology that 10 percent of the equipment used was specifically designed to eliminate it.

Signal-to-Noise Ratio

As with anything having to do with electronics there is a little math involved so bear with me. Most of the math is fairly simple, however, about at a high school level.

This is a photo of the first official use of the SIGSALY system on July 15, 1943. The conference call took place in an office located at the Pentagon. Shown are Lt. Gen. J. T. McNarney, deputy chief of staff, at right at the head of the table; Dr. O. E. Buckley, president of Bell Telephone Laboratories, at left end of the table; and Lt. Gen. Brehon Somervell, commanding general, Army Services Forces, using the phone while others listen on headsets. (Photo from the National Archives)

The ratio of signal to noise in an electronic device is usually measured in decibels (dB), which is based on logarithmic scale. Logarithmic scales are very useful to describe such ratios, particularly when large ranges of values are being measured which would be difficult to express using ordinary numbers.

Mathematically the signal-to-noise ratio (or S/N) is expressed as:

$$S/N = 20 \log_{10}(V_s/V_n)$$

Where V = signal strength measured in microvolts, V_s = signal level, and V_n = noise level measured in microvolts, with the result (S/N) measured in dB.

To illustrate how that ratio works, if $V_s = V_n$ then the result is S/N = 0 dB. There would be no intelligence conveyed in that signal since the noise would completely nullify it. More importantly, this would remain the case whether you started with 1 microvolt or 100 microvolts of energy.

As long as V_s is the same as V_n they will neutralize each other, along with the intelligence in the signal. However, if the signal were raised from 0 to 10 microvolts (as measured in an electrical circuit) and the noise remained at 1 microvolt, the result would be:

$$S/N = 20 \log_{10}(10.0) = 20.0 \text{ dB}$$

So any strategy that increases the signal voltage over the noise voltage will result in an increasingly positive S/N ratio.

To demonstrate:

3.01 dB = 2 times greater than 1 dB
10 dB = 10 greater
20 dB = 100 greater
30 dB = 1000 greater

An investment strategy that increases the S/N ratio from 10 dB to 20 dB will be rewarded by a hundredfold increase in intelligence. If you increase the S/N ratio from 20 dB to 30 dB, the payback will be *1,000* times more intelligence in the signal. As you will see, analog methods simply do not provide the same payback because of the effort required in making such improvements.

Limits Of Analog Technology

Once it was understood how noise affects signals, the engineering community involved in radio design turned its attention to improving radio through various investment strategies. Investment strategies used to increase the S/N ratio depend on the resources (time, money, skill) available in specific cases.

"An investment strategy that increases the S/N ratio from 10 dB to 20 dB will be rewarded by a hundredfold increase in intelligence."

In the very early days of radio, before we understood much about tuned circuits or even how radiowaves propagated, the primary investment strategy was to simply increase the power in the transmitter. As a result, huge spark-gap transmitters that used 1 million to 2 million watts of power were used to overcome the limitations of early receiver designs, which depended on tuned circuits, rather than vacuum tubes, for amplifying signals. This brute force method was not only expensive, it didn't guarantee a good S/N ratio at the receiver end—even under the best conditions. Designers then began investing in better antennas, improved methods of tuning, increased signal amplification (particularly with the introduction of vacuum tubes), and the creation of efficient diode detectors.

The most significant improvement, however, came with the invention in 1913 of the regenerative radio by Edwin Armstrong, followed by his invention of the super heterodyne radio during World War I. Both designs used significantly different strategies to achieve the goal of *boosting the signal voltage through amplification for an improved S/N ratio in the receiver.* That goal was achieved by either controlling the feedback in regeneration or by changing an input frequency to an intermediate frequency, then amplifying it.

The only drawback to this early method of improving the S/N ratio was the analog signal itself. In any analog circuit the degree of improvement in S/N ratio depended entirely on the strength of the original signal. That's because the original signal is always used to "drive" the final output of the analog circuit, where amplification or processing always includes some degree of added noise. A very weak signal will not "drive" the output to a level high enough to overcome the noise that is added during amplification and processing, no matter what measures are taken to suppress that noise.

This was especially evident in early vacuum tube technology, when you could actually hear the sound of the electrons rushing through the tube as you increased the audio volume, even without an external signal present in the circuit. Even today, some people still like to use "hollow-state" radios—it's hard to forget the "Niagara Falls roar" of a fully cranked tube amplifier when you're trying to dig a weak signal out of the background noise!

To give those of you who haven't worked with vacuum tubes an idea of what was inside SIGSALY, here's a photo of a model 2051 Gas Tetrode Thyratron vacuum tube. There were 384 of these inside of one terminal. Imagine how hot the equipment must have gotten when they were all working!

When solid-state devices, such as transistor amplifiers, were finally introduced people expected that noise to be reduced, if not eliminated. However, it turned out that even in the *best* designed solid-state circuits it was hard to reduce background noise entirely because of thermal changes taking place within the crystal structure of the silicon (and other semiconductor material) used in those early transistors. It was ultimately discovered that, in regard to processing analog signals, the more complex the technology, the more time, money, and energy had to be invested in support strategies to keep that technology working. Remember those SIGSALY efforts!

You can see a mock-up of a SIGSALY terminal at the National Cryptologic Museum, located adjacent to the National Security Agency's headquarters at Ft. Meade, Maryland. The museum is open to the public and free. Its collection is one of the finest in the world and contains artifacts dating back to the American Civil War. Check out its website at www.nsa.gov/museum/index.cfm for more information. (Photo courtesy National Security Agency)

As SIGSALY and other digital projects from the age of vacuum tube and transistor technology have shown, you *can* make analog technology support digital signals, but only when you have a support strategy in place that will allow it to work properly. However, analog technology suffers from the law of diminishing returns, and with each layer of complexity you add to your analog device you will experience increasingly greater levels of noise which become increasingly expensive to eliminate. The noise can always be traced back to one source: the traditional components (coils, resistors, capacitors and amplifiers) employed in processing an analog signal. No matter how well built those components are, their values and performance will always shift in and out of ideal settings because of environmental factors like temperature and vibration.

You're left with two choices: either you continue to invest in expensive technological strategies that will provide only marginal returns when applied to current analog technology, or you can try something entirely new.

Digital Strategies And Their Benefits

When it comes to new ideas about investment of time, money, and effort, sometimes it's the idea that appears to be the craziest that work the best. Consider the dilemma with analog circuits, where the components *themselves* are the cause of the problem (noise) that you're trying to eliminate.

For instance, what would you say to someone who comes along and says, "Well if the problem is the components, why not get rid of them?"

That may appear, at first glance, to be a nutty suggestion, but is it really that crazy? Consider what the definition of true craziness: doing the same thing over and over and expecting a different result.

So what's crazier? Using outdated analog technology and expecting it to perform miracles when it's obviously obsolete? Or moving on to true digital technology? Continuing to use the same old methods found in analog radio technology is just plain nuts when you know that the outcome will always be increasingly diminished returns on your investment.

This is where switching to digital technology makes sense. Since the advent of inexpensive digital technology in the 1980s and '90s, the improvement in the rate of return on investment, particularly when it comes to new strategies for increasing S/N ratio, has been significant. Each time you invest in a digital circuit design your results will be closer to an ideal solution than you can achieve with an analog circuit. This is because of three key characteristics of digitized circuitry: they can be operated at close to perfect tolerances; they do not depend upon a signal to "drive" signal processing; and they will never introduce any external noise when they process a signal.

What's more important is that when you use digital technology you literally *eliminate* circuits, thereby eliminating the problem inherent in such circuits. Certainly there are electronic circuits involved in digital technology, but they're not directly involved in the processing of radio signals. Instead, the electronic circuits are now used to simply transfer the digital information to a digital processor, where it is acted upon, and transferred back.

Furthermore, since the information is digital, there is absolutely no opportunity for noise to be introduced during the transfer. The key point is that because of digital technology's simplicity it's possible to achieve very large returns for relatively small investments of time and resources.

Next Month

Next month I'll take a close look at how traditional circuits have been eliminated through analog signal sampling using digital sampling devices. I'll show you how those digital signals are then processed with DSP software and then processed back into analog signals. Then we'll examine how DSP technology uses mathematical modeling to process digital signals more effectively than traditional coils and capacitors.

You can e-mail me with any questions at carm_popcomm@hotmail.com. While I can't answer general questions on computers, I'll be more than happy to help you with any issues raised in the columns.

As I mentioned last month, some people in the weather office are predicting up to nine significant hurricanes this summer, which is disquieting given that there are still many people who feeling the effects of other natural disasters. I would like to suggest that rather than waiting for a disaster to occur before contributing, you send a donation now to the American Red Cross (www.redcross.org/donate/donate.html) to help your fellow Americans in this time of trouble. However, there are many other good (and ethical) organizations that you can contribute to, so please use them if you wish, just don't give into "charity fatigue." If you have a job, a family around you, and live in a stable neighborhood, then show your thanks for that wonderful good luck by sharing with someone less fortunate, and do so regularly.

Let us also not forget our troops overseas who continue to need our visible support, particularly as tensions in the Middle East and Afghanistan continue to rise. Please refer to the U.S. Department of Defense's official web page, "Defend America." There's a section found at www.defendamerica.mil/support_troops.html with an amazingly wide range of practical and useful ways that you can directly help. If you are fortunate to live in the United States of America, please remember to give thanks for your personal blessings by remembering to pass on that blessing to others through regular acts of selfless sharing. See you again next month ■

by Joe Cooper, carm_popcomm@hotmail.com

Digital Signal Processing—Part III
The Haunted Father Of Digital Technology

As I pointed out in last month's column on digital signal processing (DSP), the theory behind today's "modern" technology is far from new. In fact, today's electronic DSP technology has its origins in the very early days of telegraphy. Remember, telegraphy (the original digital mode) was first employed as a purely visual means of communications, using various systems of semaphores, which could employ arms, flags, suspended balls, or sign boards.

This medium of visual communication was a vast improvement over earlier systems such as couriers on foot or horseback, but the signal (the visual images of the semaphores) could be interfered with by smoke, fog, rain, or nighttime, thus losing the message (or intelligence). Just as today's radio signals can be interfered with by noise (static, electrical discharge, or lightning strikes), or fading (disturbances in the ionosphere), those visual telegraph "signals" could encounter their own form of noise with visual interference.

One can argue that the act of converting a visual telegraphic signal into electrical pulses and transmitting them along metal wires was the first true application of electronic DSP technology. The use of such electrical signals over wires was indeed a successful strategy for reducing the amount of noise in a given signal and to ensure that the intelligence in a given signal was delivered.

Electrical telegraphy is one of the most robust forms ever developed for the transmission of information over a long distance and has profoundly influenced the development of today's digital technology. Your computer, CD player, digital cell phone, HDTV, and a host of other devices are in a sense nothing more than glorified telegraph keys, wires, and sounders.

In the 1930s, thanks to a desire to incorporate that robustness into the transmission of audio signals (whether through wires or radio waves), the theoretical foundation of the digital revolution was established. The major breakthrough came about when a brilliant, but eccentric, English scientist attempted to build a machine to communicate with the spirits of the dead, using a combination of spiritualist ideas and scientific theory. In fact, he claimed that many of his ideas for digital communications came from discussions with the spirit of 19th-Century scientist Michael Faraday. In addition to conducting these "collaborations," he spent time trying to scientifically measure "moon beams" and to communicate with an American Indian "spirit guide" named Red Cloud.

Despite his eccentricities, however, this British scientist was at the forefront of the development of single sideband (SSB) modulation and its use in long distance communications in the shortwave bands. Such early theoretical work led to our current methods of digital sampling, which is the core of DSP technology.

Alec Harley Reeves

When voice-based radio broadcasting became popular in North America during the early 1920s, AM and analog circuitry was already old hat. Early radio broadcasting used what was then 40-year-old telephone technology in its microphone and audio amplifying components.

By the time radio broadcasting and point-to-point transmissions matured in the 1930s, it was apparent that AM, and even SSB transmission, had reached its operational limits.

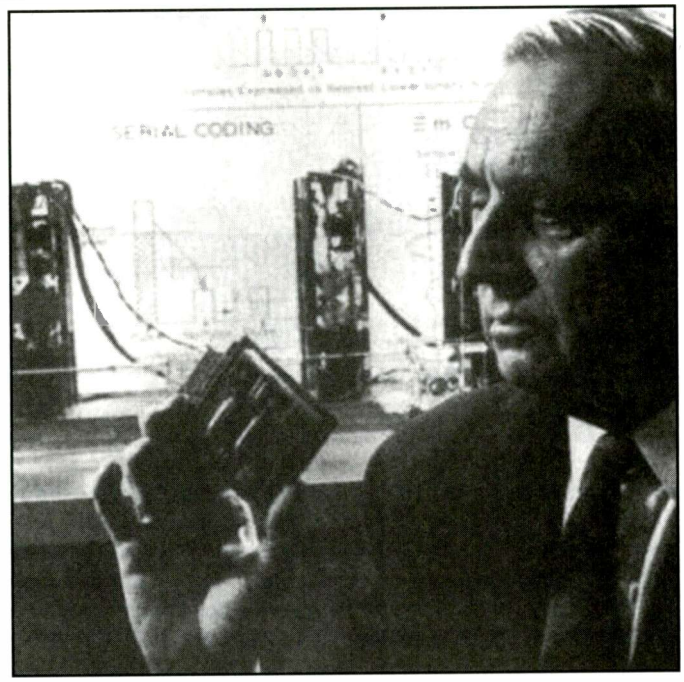
It was during this pioneering period that our British scientist and engineer, Alec Harley Reeves (1902–1971), began to look beyond the limits of conventional methods of modulating radio signals. If Reeves's name is unfamiliar to you, it is a great shame since he almost single-handedly changed the course of 20th Century history with his influence on radio engineering theory and practice. It's possible that few people know about Reeves because his work was also the foundation for several very important top secret military projects that were, in many respects, as significant as the atomic bomb. One such project was the SIGSALY system of encrypted communication, described in last month's column, which relied on Reeves's digital theories.

Prior to that important project, Reeves assisted in the development of the High Frequency Direction Finder (also known as HF/DF or "Huff Duff"), which was used to triangulate the position of enemy submarines when they transmitted a radio signal. Likewise, HF/DF could triangulate the location of British fighter squadrons, particularly important when they were out of range of the early RADAR systems.

The HF/DF project led to Reeves's direct involvement in developing a system of radio-enabled pinpoint bombing, called "OBOE," for the British Air Force during WWII. This system allowed bombers to track two microwave signals to their convergence point, which was set to a military target. Bombers flying at 30,000 feet could then deliver payloads within 50 yards of their targets, enabling the British to eliminate virtually of all Germany's manufacturing and transportation capability. The OBOE military targeting system was so precise that it was not superseded until the introduction in the 1970s of GPS satellite positioning.

Reeves was also the inventor of fiber optic transmission and led the development team that created the first practical fiber optic system in the late 1960s, which he predicted would eventually allow people to work from home using a linked computer network, foreshadowing the Internet. And to top it all off, he also perfected the condenser microphone, the cornerstone of today's recording and broadcast industry.

Each one of these inventions alone would have been a significant lifetime achievement for any one individual, and this list only *highlights* Reeves's long list of accomplishments, which resulted in over 100 unique patents held in his name. For many, however, Reeves's invention of the digitally based pulse code modulation (PCM) in the late 1930s, which is the basis of all digital technology, was his greatest accomplishment. Ironically, the foundation for his theory of PCM came out of his research into communicating with the spirits of the deceased using electronic equipment he designed and built himself. Reeves's story shows again how truth is often far stranger than fiction.



Alec Reeves, the father of the information age, was a truly brilliant scientist and engineer, yet eccentric in his personal beliefs. The CD player, computer sound card, digital cell phone, and any number of other digital sound and video recording and playback devices are the results of Reeves's revolutionary concepts. His theories developed from his attempts to contact the spirits of the departed through electronic digital sampling. (Photo courtesy David Robertson)

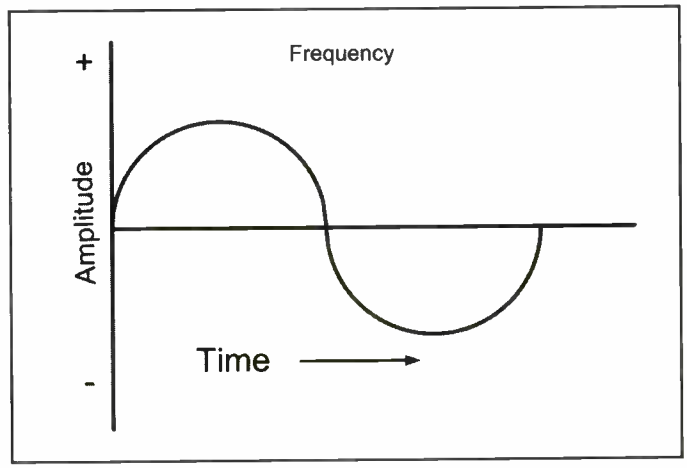
ESP And The Digital Revolution

Educated at the Imperial College of London, Reeves first found employment in 1923 with Bell Lab's International Western Electric, at the time was one of the world's leading manufacturers of radio equipment. In 1925 Bell Lab's parent, AT&T, sold International Western Electric to International Telephone and Telegraph, an independent European holding company that specialized in acquiring telecommunications companies around the globe. Reeves went to work in ITT's laboratory in Paris.

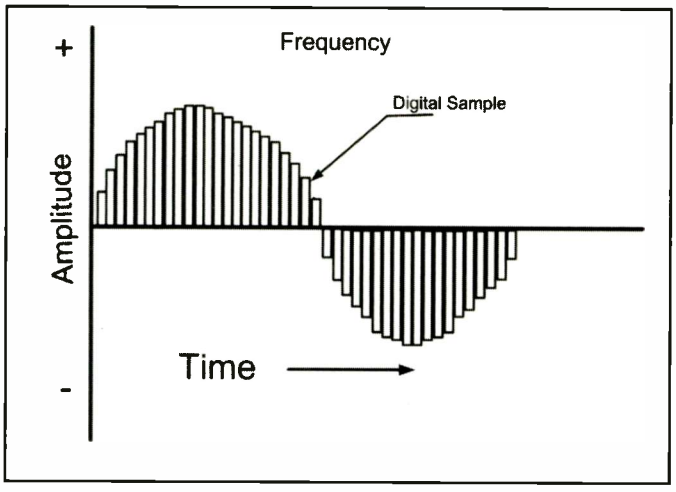
While a respected scientist during working hours, his personal time revealed his eccentric side. Reeves made no secret of the fact that his real interest in life was pursuing spiritualism and paranormal phenomena. After his death in 1971, biographers who examined his private papers found extensive notes on conversations with the spirit of Red Cloud (1822–1909), a Lakota Chief who successfully defeated the U.S. Army in Wyoming and Montana in the late 1860s. In addition to having Red Cloud as a personal spirit guide, Reeves also made claims to friends and colleges that he was in almost constant communications with the spirit of eminent 18th Century scientist Michael Faraday. Not only did Reeves claim contact with Faraday, he believed Faraday was giving him instructions on how to create his inventions.

Through his random "spirit channeling" experiences, Reeves became convinced that it was possible to communicate with the spirit world in a *rational* way, such as with telephone, telegraph, or radio. As a result, Reeves spent a considerable amount of time in developing radically new technologies in an attempt to achieve that goal. Reeves believed that any inter-dimensional communications would have to take place through something equivalent to Morse code and so became an expert in all aspects of digital communications.

Reeves developed an electronic "ESP machine" with which to communicate with departed spirits. His machine could generate a random sequence of Morse code characters and then apply a "decoding" methodology to the sequence with the goal



This is a standard analog signal showing its characteristic sine wave. The two main observable characteristics are amplitude (how loud it is) and frequency (the number of plus and minus oscillations it contains over a measured period of time). Less apparent is the fact that such a signal contains a continuous stream of information. In the real world, this signal would also carry noise, limiting its usefulness in communicating information effectively.



This is a graphic representation of the analog signal after it has been digitally sampled. In much the same way a camera takes a series of still photographs to create a "motion picture," each section of the analog signal has been analyzed and recorded in digital form. Each sample point is represented by a square, and each square contains a unique set of digital information.

of finding meaningful "intelligence." In the hands of a mere crank this undertaking would have been simply an insane project. However, Reeves was far insane and his work had a real and rational application.

Today, with computers and other digital devices, generating Morse code randomly is easy, but all Reeves had to work with were switches, relays, and other primitive electronic devices. Despite the difficulty involved, he built the machine and made it work; whetherhe received any messages is open to debate.

Reeves also built a working electronic television that he hoped would enable him to view inter-dimensional images. While the intent of using ESP devices to view extra-dimensional events was questionable, the technology Reeves was creating was rational. With these ESP devices, Reeves attempted to create a digital "sample" of extra-dimensional phenomena and then convert it to an analog form so it could be viewed in "our" world. The technology behind Reeves's ESP machine? You guessed it, DSP.

DSP allows you to create a digital sample of something in the "real" world, convert it into digital form, and then convert the generated digital information back into analog form so the information can be used in a practical way. When Reeves turned his mind to creating a new mode of digital transmission in the late 1930s, he simply applied his research into paranormal phenomena to a practical application.

The Limits Of SSB

Reeves's research in digital radio communication grew out of his pioneering work in SSB radio. The United States Navy developed SSB transmission during experiments that focused on improving the efficiency of voice communication in an increasingly congested radio spectrum.

The Navy tested various new methods of voice transmission at their experimental radio station in Arlington, Virginia, in 1914. In these experiments Navy engineers found that by suppressing the carrier and one sideband in an AM signal, the signal-to-noise (S/N) ratio of the received signal was significantly improved. Much of the Navy's radio communications took place at very low frequencies (10 to 50 kHz) where bandwidth was very limited. Because SSB modulation removed roughly half the bandwidth required to transmit a signal, a larger number of signals could occupy a smaller amount of RF "real estate," and do so with greater efficiency.

In 1915, however, John R. Carson of Bell Labs applied for a patent on SSB technology, which resulted in prolonged litigation over who actually owned the intellectual rights to this method of transmission. Carson and Bell Labs eventually won the patent rights, in 1923, after which IT&T farmed out further development work to ITT in Europe.

Bell Labs used ITT, rather than its own research staff, because of Reeves's expertise of in high-frequency radio, such as shortwave (2 to 30 MHz) and VHF (above 30 MHz). AT&T wanted to use its newly patented SSB technology to establish a network of long-distance radio telephone links around the world, and Reeves was placed in charge of a research team tasked with applying SSB transmission technology to shortwave frequencies. At that time, scientists were just beginning to understand the phenomenon of long-distance skip propagation on shortwave frequencies, and this research was as leading edge as satellite communications is today. Reeves and his team established the first practical radiotelephone link between France and England, then to the United States.

While this newly developed mode of transmission was a great improvement, Reeves came to realize that even this accomplishment was ultimately limited because of the inherent limitations of SSB to deliver an ideal S/N ratio, even in the best transmission and reception conditions. The reason was simple: any mode of communications that employed an analog method of modulation would always amplify noise and errors equally with the signal. Reeves eventually proposed a radically new method of audio communications where a signal would be transmitted using the same robust digital method used to transmit Morse code.

How Digital Sampling Works

During the late 1930s Reeves had a specific goal in mind with the creation of a radically new method of modulation. For Reeves that method of transmission had to be digitally based as that was the only way to produce a truly robust signal, meaning one with a near perfect S/N ratio at the receiver. Moreover, digital transmission would provide the following ideal characteristics:

- Quality of the received signal would depend *only* on the state of the original signal.
- Quality of the received signal would be independent of transmission media.
- The method of transmission would be completely compatible with different media and intelligence (video, audio, data).
- The technology involved would be low cost.
- One could embed new features in the new medium very easily.

Although Reeves set these concepts down in 1937, they're an exact description of the primary characteristics found in today's DSP and software-defined radio (SDR) technology. At the time, however, the technology Reeves envisioned was more

akin to a highly sophisticated telegraph system than today's micro-computer-based radios. Still, his genius let him see what others missed: the foundation of Morse code technology was what we now call digital sampling.

Rather than simply taking for granted the action of a telegrapher sending code and having it decoded at a receiving station, Reeves saw something more important happening—the telegraphic process was the creation of a virtual copy of the original text with an encoding process occurring in the mind of the sending telegrapher, who then converted that virtual copy into digital form with a telegraph key. That virtual copy was then reconstructed back into its original form at the receiving station through a decoding process taking place in the mind of the receiving telegrapher.

More significantly, Reeves believed that the large quantity of data required to describe complex events as digital information was possible. All that was required was replacing the human beings operating telegraphy keys with high-speed electronic devices. There was already an example of Reeves's approach—motion pictures, which are actually just digital samples.

Motion Pictures As Digital Sampling

The movie camera is simply a device that takes a sample of events occurring in the real world and converts them into photographic images on a moving roll of light-sensitive celluloid film. When light focused through a lens falls on chemically treated photographic film, the chemicals react to the light in a gradient of impressions between dark and light (using black and white film as an example just to keep things simple). That film is processed through a chemical treatment to make the photographic images visible. These can then be viewed by means of a movie projector.

The movie camera and the digital sampling device used with radio signals are similar in how the sampling is performed. The movie camera does not make a continuous record of the events occurring in front of its lens (as an *analog* tape recorder makes a continuous record of sound occurring in front of a microphone). Instead, the movie camera takes a single photograph so many times per second and does so in a sequential manner as the photographic film is pulled past the camera lens. So what you have once the film is processed is a sequential record of individual events that took place in front of the camera.

When that film is run through at projector, each individual picture is placed in front of the lens at the same per-second rate that it was photographed. Each frame stops, is projected, then replaced by the one behind it, which is replaced by the one behind it, and so on. Because of a phenomenon called "persistence of vision," the human brain perceives each of these individual frames shown in sequence as being in motion, and thus appearing to be the same state as the original.

The method of digital sampling signals (audio, video, or radio) is almost identical to that used in a movie camera. The only difference between a movie camera and a digital sampling device is that the latter takes an electronic "snapshot" of the state of that signal rather than a photographic picture. So instead of bringing a radio signal into a conventional analog circuit from an antenna, as when using a super-heterodyne radio, a digital sampling device is hooked up to that antenna instead.

Thus when the digital sampling device is turned on, it begins to take electronic samples of the signal that it finds on the antenna, and it does so much like a movie camera because it records those samples as a series of timed events. However, unlike a movie camera that may be taking 24 samples per second, the digital sampling device takes samples many millions of times per second. This raw digital information can then be processed (or, to continue the analogy, developed like the pictures on the movie film) to convert the digital information to an analog form.

A movie projector takes individual pictures and makes them "move," and a device that converts the digital information back into analog form performs a similar task. And, in the same way the movie's relatively small images can be projected (amplified) to a significantly larger size with relatively little noise (grain or distortion on the film), the same "low noise" amplification of the digital signal takes place.

What also takes place—and this is critical for a full appreciation of the strengths and weaknesses of digital sampling techniques—is a loss of information "between frames."

Reeves, however, had some interesting ideas on how to tackle this problem, each approach giving today's engineers an important foundation to work upon when developing DSP solutions.

Coming Next Month

Next month I'll finish this series on the foundations of DSP technology with a look at how Reeves envisioned a means to successfully capture large amounts of digital information and then convert it back into analog form. Although Reeves had no practical way of testing his theory at the time, when the technology did become available many decades later, his ideas turned out to be completely correct.

E-mail me with any questions at carm_popcomm@hotmail.com. As mentioned before, I cannot answer general questions on computers, but will be more than happy to help you with any issues raised in the columns.

Even though summer has just begun, early severe storms have already taken lives and left many homeless. I would like to suggest that rather than waiting for a disaster to occur before contributing, you send a donation now to the American Red Cross (www.redcross.org/donate/donate. html) to help your fellow Americans in this time of trouble. However, there are many other good (and ethical) organizations that you can contribute to, so please use them if you wish, but do not give into "charity fatigue."

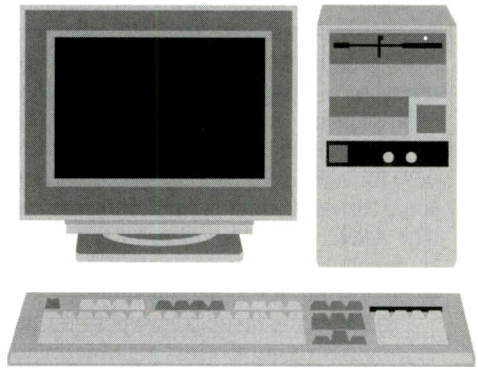
Again, if you have a job, a family around you, and live in a stable neighborhood, show your thanks for that wonderful good luck by sharing with someone less fortunate, and do so regularly.

Let's also not forget our troops overseas who continue to need our visible support, particularly as tensions in the Middle East and Afghanistan continue to rise. Refer to the U.S. Department of Defense's official webpage, "Defend America," where you can find an amazingly wide range of practical and useful ways that you can directly help (www. defendamerica.mil/support_troops.html).

If you're fortunate enough to live in the United States of America, please remember to give thanks for your personal blessings by remembering to pass on that blessing to others through regular acts of selfless sharing. ■

by Joe Cooper, carm_popcomm@hotmail.com

Digital Signal Processing—Part IV Wrapping Up Our Exploration Of Digital Sampling Technology



Over the past series of columns we've learned that digital technology is not something new. The "digital revolution" has actually been in development for just over 300 years. More importantly, you now know that the "ancestor" of any digital technology that may you own—CD player, cell phone, digital signal processing (DSP)-based radio, or even a micro-computer—was a telegraph key. All of today's digital technology can be traced back to the first practical telegraphy system developed by Claude Chappe in France in 1792. And the first true application of DSP was when Baron Pavel Lvovitch Schilling began transmitting telegraph signals in 1832 using electrical wires.

Edwin Howard Armstrong (1890–1954), inventor of the regenerative, super-regenerative, super-heterodyne, and FM radio. His was a life of brilliant invention and achievement, but hounded by lawsuits from competitors. Few people on this planet have been untouched by his accomplishments, but other than in the circles of those interested in radio history and technology, his name is virtually unknown. (Photo courtesy NOAA)

Even those critical improvements made in telegraphic technology, such as American scientist Joseph Henry developing the mechanical striker, were simply further applications of DSP technology. One could even argue that when Samuel Morse developed his alphanumeric code to exploit the commercial potential of Henry's mechanical striker he developed the human equivalent of a computer software program to encode and decode a message.

Telegraphy, whether visual, over wires, or wireless, has always worked exceptionally well as a method of communication, so much so that the radio engineering community of the early 20th Century agreed that a method was needed to allow voice and data communications to be transmitted in digital form.

The Beginnings Of Analog Radio

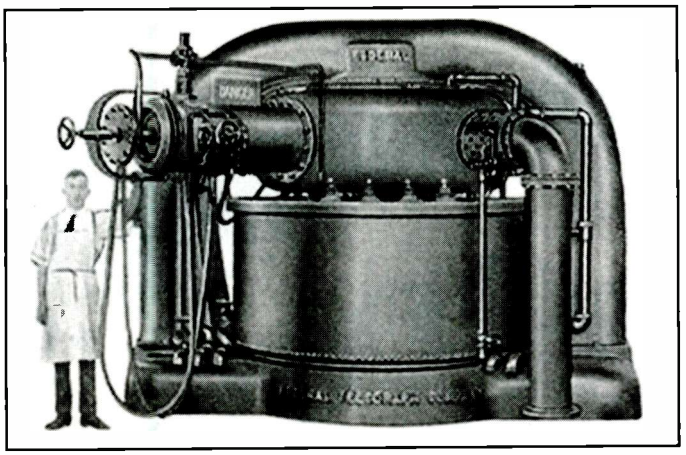
When early attempts were made at analog voice communication, the techniques first used were already obsolete. The carbon microphone used in the first successful voice broadcast by Reginald Fessenden in 1906 was already a relic of 18th Century technology. However, despite the limitations involved the exciting potential for the communication of voice and music motivated amateur radio experimenter to use whatever was on hand. What most hams used when attempting early voice communication was the carbon microphone, which was relatively cheap and easy to find. These early amateur-built radios and transmitters were crude, even by the standards of the day, but they worked.

More importantly, people of the day were willing to ignore the technical limitations of the early radios simply because they were so entranced by the novelty of the new medium. In fact they were *so* willing to put up with these limitations that a completely new hobby came into being almost overnight. People spent hours listening to noise-filled signals looking for new stations, and soon these early radio listeners began to log the stations they heard. And with skill and patience they began to hear new stations at greater and greater distances.

While there were many amateurs on the air, one of the more interesting stories concerns one Dr. Frank Conrad, assistant chief engineer of Pittsburgh's Westinghouse Electric Company. Conrad became interested in radio in 1912 and received a license (callsign 8KX) to transmit in 1916. He spent the next few years as an amateur broadcasting from a garage in his backyard.

Much to his amazement he was soon deluged with requests for more music and information, which Conrad soon turned into a regular broadcast schedule, using musical records supplied by a local music store to supplement news and talks. After several years of performing these broadcasts, Conrad's boss, Westinghouse vice president H.P. Davis, saw a newspaper ad stating that a Pittsburgh department store was selling a radio receiver that could pick up Frank's broadcasts. This advertisement convinced Davis that there were enough listeners for Westinghouse to invest in commercial broadcasting.

In 1920 the Westinghouse Corporation established KDKA in Pittsburgh using amplitude modulation (AM) as the method of transmission. At the time this choice made sense because analog signals could be received using simple radio circuits, including those in crystal radios



This is a 1000-kW Poulsen arch transmitter, manufactured by the Federal Telegraph Company after World War I and installed at the Lafayette radio station. It was the only arc transmitter of its time that could produce a continuous wave. This required a unique method of keying, which employed two shifting frequencies. Frequency shift keying (FSK) was the starting point for the development of FM radio technology by Armstrong in the 1930s.

which could be built by anyone with simple tools and relatively inexpensive parts. As a result, the original group of amateur radio monitors, who numbered in the hundreds, mushroomed into tens of thousands by 1922.

While still a fad, there were enough listeners to encourage a flock of advertisers to purchase on-air time for their announcements, putting surprising amounts of money into the coffers of broadcast station owners. This encouraged a huge boom in the manufacturing of mass-produced radios to expand the listening market command higher advertising rates. By the mid 1920s people who had no real interest in radio technology, but wanted to listen to what everyone else was talking about bought massed produced radios by the millions.

And as they say, the rest is history.

The Limitation Of Analog Radio

The large radio audience that emerged by the late 1920s was a great financial boon to the commercial interests who owned fledgling radio stations. However, the radio transmitters and receivers of the period were the bane of radio engineers who had to design them.

If you look at the research of the engineering community of that time, you'll find that most efforts focused on developing alternative methods of radio transmission and reception. AM signals and analog circuits may have been a relatively cheap and easy way to build a pioneering broadcast industry, but those who needed clean and dependable signals to do business found analog technology deplorable. That, of course, is because whenever you amplify an analog signal, you amplify noise and intelligence equally. Worse, not only was natural noise amplified, but all methods of amplification (vacuum tube and solid-state devices) actually added *additional* noise to radio circuits, and it was expensive to eliminate it once it was in there.

While there was indeed a "golden age" of AM radio research and development during the 1920s, it was over by the time the Great Depression came along in the 1930s. Even with the restraints on research during the Depression, by the end of it scientists knew that any further development of analog technology was a dead-end pursuit. The only reason why time, money, and effort was still being invested in analog technology was because commercial broadcasters were still getting rich off it and were in no hurry to kill the golden goose.

Dissenting voices among the engineering community were becoming louder, however. One of the loudest belonged to a great pioneer in the development of analog radio, Edwin H. Armstrong. Armstrong's discoveries, such as the regenerative and heterodyne radio circuit, completely transformed the way in which radios were built and used. His amplification circuits, particularly with the super-heterodyne radio, eliminated the multiple steps that were initially required just to tune a radio.

Early methods of tuning were complicated because a radio operator had to tune through a series of circuits to receive a signal on a single frequency. One reason logging a station was so important was to keep track of the multiple settings needed to find that station again. This tuning approach was fine for commercial operations, where one would monitor a single frequency (or more appropriately for those days, a particular wavelength) to listen to a small number of stations. It was unacceptable for broadcast reception, however, where one would be tuning over a wide range of frequencies to listen to multiple stations.

The beauty of Armstrong's super-heterodyne radio lay in its simplified design that reduced the multiple controls to two: the tuning dial and sound/volume control. Some visionaries, such as RCA's David Sarnoff, envisioned a time when super-heterodyne radios would have their tuning dials completely eliminated, replacing them with push buttons for selecting pre-tuned radio stations.

Even though Armstrong revolutionized radio receiver design, he also recognized the limits of analog methods of signal modulation. He knew that while his advancement was important, the circuits were still exceptionally vulnerable to noise, while digital methods would be vastly superior. There was no reason, he believed, to continue to further develop AM radio when a better alternative method could be developed.

His practical solution, which had the value of being able to use the vacuum tube technology of the day, was frequency modulation (FM), which he patented in 1933. But what many people don't know is that this mode of communication actually had its theoretical origins in the spark-gap based radiotelegraphy technology of the World War I era and earlier.

Frequency Shift Keying And FM Radio

Armstrong's inspiration for FM was the Poulsen arc transmitter, invented by Valdemar Poulsen of Denmark in 1902. Poulsen was a successful inventor with many accomplishments to his name, including the first magnetic voice recording device, developed in 1898 and incorporated into the first commercial telephone-answering machine the following year. Unlike the conventional spark gap transmitter of the time, which produced inefficient dampened waves of radio frequency, Poulsen's transmitter produced continuous waves. It was not only more efficient than the spark gap transmitter, but significantly smaller as well. Not surprisingly, the United States Navy adopted the Poulsen transmitter in 1912 and installed them at its major transmitting sites, as well as onboard many of its important battleships.

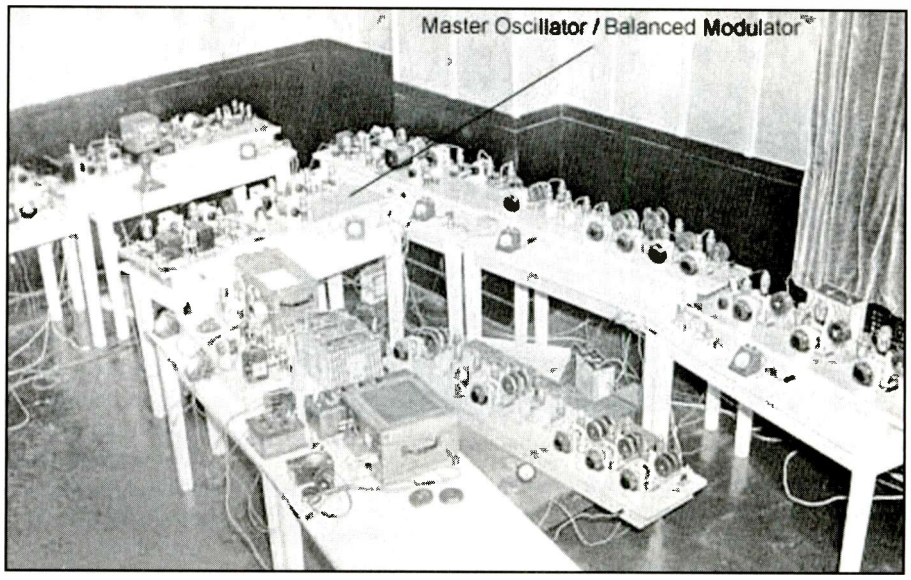
What caught Armstrong's imagination was the method of

keying the Poulsen transmitter employed, which was to shift the frequency between two points. Even a few years after the Poulsen transmitter was put into use, radio engineers were seriously asking if frequency shift keying (FSK) could sample an audio signal and convert it into digital form for transmission. It seemed possible if the simple "on" and "off" method of FSK was used in a Poulsen transmitter for radio telegraphy could be controlled by voice. Serious efforts were made to substitute a microphone for a telegraph key, resulting in the FSK-based signal being modulated by a voice or music. Likewise, it also seemed being possible to design a radio receiver that could convert the FSK signal back into analog form.

What's being described in this method of transmission is digital sampling, though at that time the concept was not fully understood. While exploring the theoretical concept of how FSK could be used for voice transmissions, the engineers of the period (early 1920s) could not believe what they discovered. When the mathematical calculations were performed, they confirmed that the bandwidth required to make audio sampling work would be double the highest sampled frequency. This is exactly the same conclusion Harry Nyquist would outline in his famous theorem on digital sampling in landline telegraphy, published in 1927.

The problem was that at the time most radio transmissions were being made on frequencies below 1.5 MHz, so the bandwidth requirements of FM were unthinkable. However, Armstrong's solution for the large amounts of radio frequency real estate required was to simply move FM transmission far above the 1.5 MHz into the VHF portion of the radio frequency spectrum. In this part of the spectrum, Armstrong argued, one would be able to operate FM transmitters at their optimal bandwidth for good quality sound. Indeed if you take a look at the current band-plan for commercial FM broadcasting in the United States, you find that each station is allocated 200 kHz.

Thanks to an important discovery Armstrong made during the 1940s, audio transmissions were not the only signal transmitted in that allocated bandwidth. He found that a single FM carrier wave could transmit two radio programs at once. He was even able to simultaneously transmit voice, a telegraph message, and a facsimile of the front page of *The New York Times*. This clearly showed the potential of digital signals carry multiple data streams over one signal—a clear improvement over AM technology.



This 1933 photo shows Armstrong's early "breadboard" FM transmitter, which was located in the RCA radio laboratory on the 85th floor of the Empire State Building in New York City. The room was originally used for early mechanical television experiments, which were put on hold during the Depression. Armstrong would later be ordered to remove the equipment so RCA engineers could begin testing a working electronic television system. (Courtesy the Harry W. Houck collection)

FM stereo broadcasting was introduced in 1960, and there have been a few other technical enhancements, but FM proved itself an extremely successful medium of broadcast almost from the very beginning.

Opposition To FM By Big Business

Given the obvious superiority of the digitally based FM over analog-based AM, why did AM broadcasting and its associated technology remain so firmly entrenched for so long? It took over 40 years for FM broadcasting listeners to surpass AM audience's numbers in the United States and Canada, while most of the world's other broadcasters and audiences embraced the new mode soon after its introduction in the 1940s.

The consensus among radio historians today is that this delay was caused by the deliberate resistance of David Sarnoff in his role as president of RCA during the 1930s and '40s. Armstrong had sought Sarnoff's assistance in developing FM, but Sarnoff was not particularly keen to support him. Sarnoff knew that AM radio had reached its limits of development, but rather than explore a new mode of audio transmission, he believed that video was the mode of the future. Sarnoff and RCA had been experimenting with mechanical television systems in the early 1930s, using a research facility located in New York's Empire State Building. RCA engineers had converted an abandoned mooring tower for dirigibles located on the top of the building into a mast for VHF antennas and installed a transmitter below it.

In 1933, during the height of the Depression, Armstrong approached RCA for help in developing FM radio, and Sarnoff did allow him to use the Empire State facilities for that purpose. Armstrong set up his equipment in the spring of 1934, operating experimental station W2XF on 41 MHz. But one year later, Sarnoff ordered Armstrong to vacate the facility without any explanation offered (it was actually for the resumption of television experiments).

Sarnoff could not have cared less about Armstrong's FM experiments, which he considered purely academic. Armstrong, though, was fully determined to make FM the new mode of broadcasting in the United States, if not the world. This was not out of an altruistic desire to see the state of radio improved for the listener, but with the purely commercial aim of receiving a patent royalty on every receiver and transmitter sold. Armstrong's shot across the bow of the AM broadcasting establishment came in 1936 when he pub-

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The antenna for the United States' first commercial FM broadcaster was located on Mt. Asnebumskit near Paxton, Massachusetts, and was completed in 1938. The station operated as W1XOJ on 43 MHz at 50,000 watts until the FCC ordered all FM broadcasters to move to the current FM band in 1945. The tower remains in operation to this day. It's 400 feet high and sits at 1,400 feet above sea level, providing a very good signal to the Boston area. The transmitter was originally linked to the studio 42 miles away by a repeater transmitter operating at 133 MHz. After 1945 the station converted its transmitter and antenna to the higher frequencies and operated under the callsign WAAB, which it later changed to WAAF in 1967. (Photo courtesy www.tybush.com)

lished his now classic paper on FM broadcasting, titled "A method of reducing disturbances in radio signaling by a system of frequency modulation" (https://michael.katzmann.name/armstrong.html). With this, Armstrong claimed the future of FM radio as his own.

In that same year, Armstrong approached the FCC to consider laying the foundations for FM broadcasting, which was allowed through the establishment of a number of experimental stations. By 1937 the FCC was satisfied that the necessary technical research had been undertaken and opened up seventy-five 40-kHz channels between the frequencies of 41.02 to 43.98 MHz. During this period Armstrong found a champion for his new mode of broadcast in the person of John Sheppard, president of New England's Yankee Network of radio stations.

Sheppard, who formed his network during the late 1920s and early '30s as a regional broadcaster, originally had no interest in connecting it with an audience larger than a single city and its suburbs. In that regard, the limited range of FM broadcasting at 40 MHz was an ideal mode of transmission for reaching such a highly defined market. But, in 1937, Sheppard received the first FCC license to build and operate a 50,000-watt FM transmitter under the call letters W1XOJ. It still operates today as WAAF. On May 27, 1939, the station officially began broadcasting from Paxton, Massachusetts, a suburb of Boston, becoming the first in the United States to do so (though 11 *experimental* stations, including one owned and operated by Armstrong, were also on the air). As soon as the official broadcasts were underway, the FCC received over 500 applications for FM station licenses in the first year alone.

During this time, Sarnoff was completely embroiled in the official roll out of NBC's system of television. However, soon after the inaugural TV broadcast made from the 1939 World's Fair in New York City, Sarnoff quickly caught up on the news of broadcasts success. He was completely furious with the development, for he had completely misunderstood what Armstrong had been attempting to accomplish during his experiments in the Empire State building.

Rather than a radically new method of broadcasting, Sarnoff believed that Armstrong was simply building a new anti-static filter for AM radio, and so had allowed him to go ahead. Now Sarnoff found that not only had Armstrong been successful in creating a new technology, he was also building a highly competitive broadcasting empire in NBC's own backyard. Initially Sarnoff believed that World War II would put an end to Armstrong's ambitions, but instead Armstrong further entrenched FM when he convinced the U.S. military to adopt the new mode as a standard. More importantly, Armstrong also demonstrated the advantage of using FM in RADAR technology, making it more accurate and reliable.

Having committed the majority of RCA's financial resources to the development of television, Sarnoff could not fight Armstrong directly in the rapidly emerging FM marketplace. Instead Sarnoff launched a back-handed attack on the FM broadcast industry, first by having the FCC move the frequencies from 40 MHz to their current location, arguing that the lower frequencies were needed for television channels. This not only rendered all existing FM receivers obsolete, but also required the complete rebuilding of transmitters and antennas because of the jump to a higher set of frequencies.

At the same time the FM frequencies were changed, RCA embarked on a massive promotional campaign for its new television system. Not surprisingly then, the majority of people in the New England area who had a non-operating FM radio opted for a new TV set instead. Not satisfied with simply stopping FM radio in general, however, Sarnoff attacked Armstrong directly through patent interference. Despite having virtually ignored FM technology, RCA still won the patent fight, stripping Armstrong of any right to royalties in the United States.

Brought to financial and emotional ruin by the years of litigation with RCA, Armstrong jumped to his death in 1954. His widow continued the fight and finally won in 1967, though receiving only a

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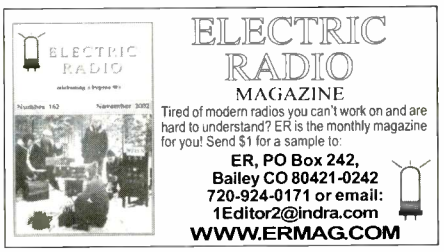
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paltry $1 million for her efforts, while RCA made hundreds of millions thanks to Armstrong's patents.

The story of Edwin Armstrong unfortunately demonstrates a cold reality of the world of business: no matter how good an idea or invention is, it has no value in the "real world" unless it can make money.

The Implication For Digital Radio

As I've pointed out with Armstrong, the scientific world originally dismissed FM as impractical because of the amount of bandwidth involved in digital sampling. Given the limited number of radio frequencies available at that time, that assessment of the situation was true.

Also, when engineers like Harry Nyquist and Alec Reeves put forward their ideas on digital sampling and its application to advance communications techniques, their ideas were critically important, but not in a way that was financially practical. Reeves realized his ideas on pulse code modulation earlier than expected thanks to the extraordinary circumstances that the Allied forces found themselves in during World War II. And while contemporary scientists fully understood and appreciated Nyquist's theorem at a *theoretical* level, they could only use it in a limited way.

However, by the late 1940s the time was right for Nyquist's ideas to be re-discovered by Claud Shannon in his paper *A Mathematical Theory of Communication*. When Shannon re-articulated Nyquist's theorem, he did so at the right time and in the right economic climate. As a result, he was able to demonstrate clearly the potential application of digital sampling in voice, video, and data communications. Thanks to concurrent advances made in areas such as solid-state electronics and DSP technology, Shannon's ideas opened up an entirely new era of communication theory.

Today all of us directly benefit from the once radical ideas of Reeves, Nyquist, Shannon, and Armstrong about DSP. The ideas of geniuses such as these, however, will always be far ahead of the times that produce them. What is important then is to make proper use of these ideas when the proper time comes along. Today is similar in the multitude of technological advances that surround us. We have common access to tools such as computers, the Internet, and integrated electronic components that would have been pure science fiction when their original theories were first articulated.

So remember, while these engineers worked separately from one another, each had a single goal in mind as they did their research. Claude Shannon best summarized this goal in his previously mentioned paper, where he began with this statement: "The fundamental problem of communication is that of reproducing at one point either exactly or approximately a message selected at another point." Over 50 years later, we are closer to reaching that ideal than at any time in history—and we are the inheritors of this amazing legacy of 20th Century engineering. It is up to us to see that the potential of this gift is fully utilized.

The reality of radio and electronics in the 21st Century is that analog technology has been obsolete for over 60 years already. It is time to put it aside and, in doing so, fully reap the benefits of living in our "new" digital age.

Next Month

Next month I'll look at the current technology used in capturing large amounts of analog information, converting it into digital form, processing it, and then successfully converting information back into analog form. There's a wide range of such converters available today and they're becoming increasingly inexpensive.

You can e-mail me with any questions at carm_popcomm@hotmail.com. As mentioned before, I cannot answer general questions on computers, but will be more than happy to help you with any issues raised in the columns.

As I write this column, it's predicted that this summer will be one of the worst on record for storms. I hope by the time you read this, prediction has not come true, but if it has, please send a donation now to the American Red Cross (www.redcross.org/donate/donate.html) to help your fellow Americans in this time of trouble. However, there are also many good (and ethical) organizations that you can contribute to, so please use them if you wish but do not give into "charity fatigue." Again, if you do have a job, a family around you and are living in a stable neighborhood, then show your thanks for that wonderful good luck by sharing of that with someone less fortunate, and do so regularly.

Let us also not forget our troops overseas who continue to need our visible support, particularly as fighting in the Middle East and Afghanistan remains unabated. Please refer to the U.S. Department of Defense's official webpage, "Defend America." The section found at www.defendamerica.mil/support_troops.html has an amazingly wide range of practical and useful ways that you can directly help.

Again, if you are fortunate enough to live in the United States of America, please remember to give thanks for your personal blessings by remembering to pass on that blessing to others through regular acts of selfless sharing. See you again next month! ■

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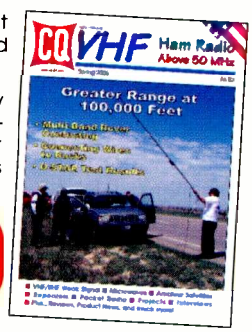

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Digital Sampling Technology: What It All Means To You

Over the past few months, I've outlined the history of digital sampling as it relates to digital signal processing (DSP) technology used in much of today's technology. While it may be hard to believe, your digital cell phone, CD player, personal computer, and software-defined radio all owe their existence to the invention of telegraphy. That's right—today's digital revolution actually began with the electrical transmission of Morse code signals in the early 19th Century.

I also showed in the recent series of columns that by the beginning of the 20th Century analog technology was seen by the engineering community as having reached its practical limits and was declared obsolete by the early 1930s. This is not that surprising as scientists, mathematicians, and engineers had been working hard throughout the 1920s to perfect a new digital method of communication.

The problem with analog is simple: it contains too much noise to be truly useful in professional communications. However, analog technology is easy to create and easy to manufacture, as illustrated by Edison's wax cylinder phonograph or crystal radio sets. Understandably, early amateur radio experimenters used inexpensive off-the-shelf analog components, such as telephone microphones and headsets, to create their first radios.

Because of the novelty of the new forms of media, the public generally overlooked the static and electrical noise that they heard in their headphones and loud speakers. In fact, our own hobby of radio monitoring began with people in the 1920s making a game of "digging out" radio signals that were "buried" in that static and noise. The reward for that game came when people began to log their "captured" stations and received confirmation of their reception in the form of letters or QSL cards.

While this was fun for the public, it was anathema to radio engineers and scientists who were trying to build profitable communications systems for the business community. Scientist Claude Shannon best articulated that goal when he published *A Mathematical Theory of Communication* in the late 1940s. According to Shannon, "The fundamental problem of communication is that of reproducing at one point either exactly or approximately a message selected at another point." But no form of analog communication achieved "exact or approximate" communication because all analog forms introduce noise into a signal.

While a great deal of effort has gone into reducing noise in analog systems, it's always a case of rapidly diminishing returns where increasingly expensive equipment produces little or no effect upon the problem. This is where digital technology shows its true value, because it can deliver "exact or approximate" reproduction. More importantly, digital transmission provides the following ideal characteristics:

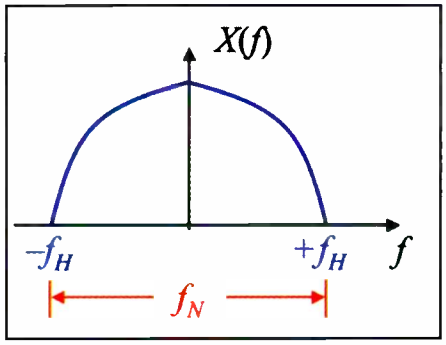
- The quality of the received signal is equal to the state of the original signal.
- The quality of the received signal is independent of transmission media.
- The method of transmission is completely compatible with different media and intelligence (video, audio, and data).
- The technology involved is low cost.
- New features can be easily embedded in the new medium.

British engineer and scientist Alec Harley Reeves first theorized those characteristics of digital communications in 1937, and they formed the basis of today's digital revolution. And thanks to the genius of other scientists and engineers, such as Harry Nyquist, Claude Shannon, and Edwin H. Armstrong, the actual technology needed to make the digital revolution work became a reality.

Digital Sampling Today

The true foundation for the digital revolution lies in the process of digital sampling. Digital sampling is a process where an analog (or continuous) signal is measured at regular intervals and those measurements are converted into digital information. This process is much the same as that used in motion pictures. There are, however, certain rules that apply for successful sampling. For instance, if too few or too many samples are taken, then what is know as sampling distortions occur. Again using the motion picture example, the phenomenon of wheels or other moving objects appearing to be going backward when a movie is projected is a visual representation of a digital sampling error.

Back in the 1920s, Harry Nyquist discovered how to prevent such sampling distortions from happening. For digital sampling to be successful, he found, one should limit the bandwidth of the signal and use a sampling frequency that is twice the bandwidth. If that formula is not followed, the result is aliasing, where different components of the sampled signals become indistinguishable from each other. Aliasing causes audio signals to sound harsh or distorted, video images to have jagged edges or moiré (interference) patterns, and radio signals to have false signals and birdies.



Here's a graphical representation of the Nyquist sampling theorem, where X(f) represents 1/2 the bandwidth fn. If X(f) balances with fn, then perfect reproduction can be achieved from a sample. However, if there is any disturbance of that balance, distortion results. The goal of the person designing a sampling device, as well as the person who operates it, is to maintain that balance as much as possible.

In this column, I'll outline the problems that can be encountered in digital sampling technology and how these have been overcome. I'll also look at how digital signals are converted back into analog form so they can be used and understood by the human eye and ear. This information is critically important for anyone using today's digital radios and accessories, including personal computers. While your radio or computer manages many of the issues involved in digital sampling automatically, you need to be able to know if your equipment is working properly.

The other issue you need to understand is that while the promise of digital technology is better reproduction of a signal, the reality can be very different. Badly engineered digital sampling can produce very bad results, often sounding or looking much worse than analog sources. Likewise, a poorly managed digital device (a radio, sound card, or video card) can result in the same thing.

As in the case of any new technology, new skills and knowledge are needed to get the most benefit from it. There will be a learning curve and time required to master the new information being provided. However, I hope that my explanations will help you to make the best use of this amazing new radio technology.

Problems In Digital Sampling

Digital sampling is the measurement of a signal at regular time intervals, with those measurements then converted into digital information. While that may appear straightforward, actually many errors can occur in the sampling process, resulting in less than accurate reproduction of a signal.

These errors mainly occur when a digital sampling device strays from ideal of Nyquist's sampling theorem in either its operation or design. Again, that theorem outlines two main criteria for sampling success: bandwidth and sampling rate. If there is a notable divergence from the ideal in either of these two critera, you'll have sampling errors, called distortions.

There are six main types of distortion that can affect digital sampling:aliasing; integration effect; jitter; noise; slew rate; quantinization; and non-linearity.

There are several minor types of distortion as well, but these can generally be controlled with either hardware or software solutions. The six main types are much more difficult to manage as they are often the result of poor design or bad component choice. More importantly, these distortions can be a result of improper settings or poorly operated equipment. Here's where the equipment operator's skill and understanding can play an important role in keeping errors from occurring. Knowing the characteristics of each problem can help the operator diagnose and eliminate the problem. So let's look at each one and see how they are experienced.

Aliasing

Aliasing is one of the most common errors found in digital sampling and occurs because a particular frequency in a signal can produce multiple sampling data due to harmonics contained in that signal. This generally occurs when sampling captures information that's outside the ideal bandwidth stated in the Nyquist sampling theorem.

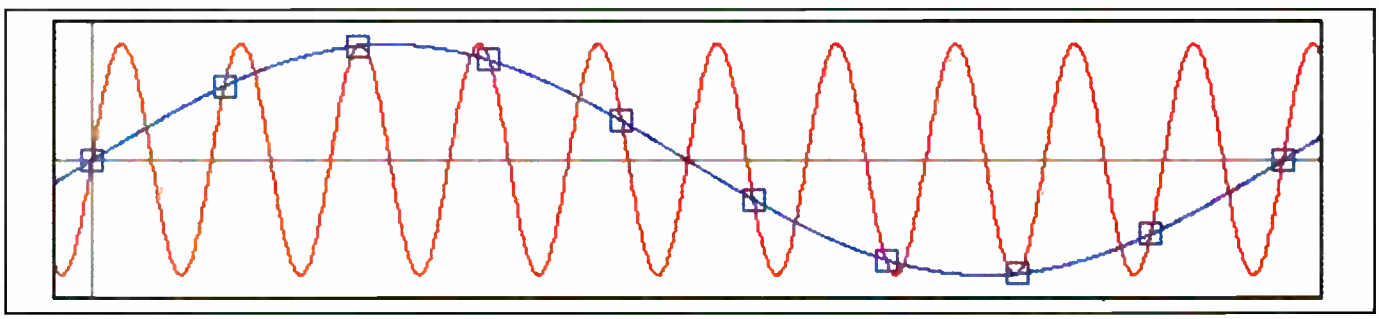
In practice, it's very difficult to achieve a precise bandwidth when sampling a signal. So rather than obtaining a single pure sample, you can end up with samples outside the desired bandwidth. When that sample is then converted back into analog form, the information outside the bandwidth is imposed upon the primary sample, causing noise.

The solution to this problem is to use a hardware or software filtering method called "anti-aliasing" to sharpen the bandwidth. In some cases these filters can be analog devices used on the input signal (such as from an antenna) before it is sampled, or DSP can be used to remove aliasing from a sampled signal.

Integration effect

This refers to the amount of time involved in the digital sample and is analogous the shutter speed of a camera and. As with a camera, setting the wrong shutter speed can result in a bad picture. In the case of a digital sample, you can have either too much or too little information. The effect can be seen in the digital or analog output, where there is noticeable distortion.

The problem can be traced to the sample and hold circuit of the sampling device where the values of the capacitors used are insufficient to match the speed required for the sample. The general solu-



An illustration of the phenomenon of aliasing in digital sampling. Shown here is a common problem: A low (blue) frequency is being sampled and the sampling produces a second higher (red) frequency as an artifact of the sampling process. This is the equivalent of wheels seeming to turn backwards in motion pictures, which is also a form of aliasing in that medium. In the case of digital sampling, aliasing can be controlled through the use of filters and other anti-aliasing strategies.

tion to this problem is to reduce the sample rate to one that the sample and hold circuit can properly manage.

Jitter

Jitter refers to an unwanted variation in the timing used in digital sampling, resulting in inconsistent timing pulses. We've all seen jitter in video signals, where the image is no longer in sync and has significant distortion. An equivalent of this jitter can be found in audio cards and DSP circuits, too, with equally disturbing distortion, generally heard as phasing. Jitter is caused by clock errors occurring during the sample, generally due to rapid variations in a signal. These variations produce phase perturbation that disrupts the timing pulse. Part of the problem is that only small disruptions of a few hertz are needed to create jitter.

The common method of preventing jitter is to use buffers to store the data to eliminate any timing errors. This is done through re-synchronizing the signal while it's held in the buffer and then releasing it when the synchronization is complete.

Noise

While digital information itself contains no noise, the natural presence of noise within electronic circuits can affect that digital information. Often circuit noise can knock out or distort a data bit, which results in the distortion, or even destruction, of the information found in that signal. So, ironically, digital circuits must be designed to have lower noise levels than analog circuits in order to work properly.

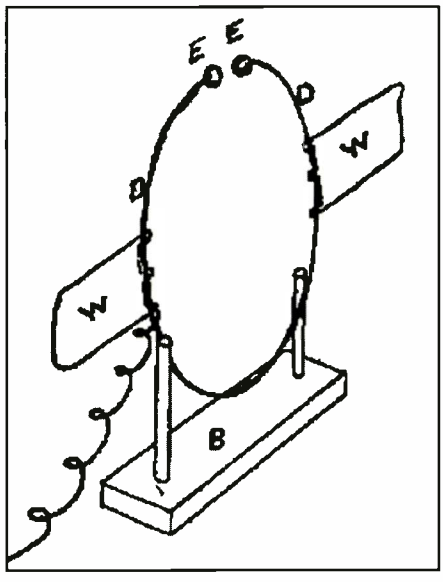
Such noise reduction designs are necessarily expensive, but the most economical approach is to build error detecting and correcting devices into the circuit. These check the data integrity of the signal and when errors occur, the information is sent again until it is correct.

In any event, no matter how much noise is present in a circuit, none is ever directly transmitted in the digital signal. That's a significant improvement over analog circuits, where any introduced noise is automatically added to the signal.

Slew rate

A slew rate error occurs when attempting to change a digital signal back to analog form. It's much the same as integration effect, except that it reflects the circuit's inability to release the digital signal either fast enough or slow enough into the analog converter circuit.

The general effect is to cause noticeable distortion in the analog signal called saturation, which is similar to over- or under-driving a conventional analog audio amplifier. The solution is to build in a limiting circuit into the digital to analog circuit so extreme peaks and troughs in the digital signal can be controlled.



This picture illustrates the receiver portion of the device Heinrich Hertz used in 1888 to demonstrate the existence of radio waves. When a radio wave strikes the metal loop, an electrical current is generated in the loop causing an electrical spark to jump across the gap. That's what happens in radio antennas today, except conventional radio receivers are used to tune to the specific frequency of one signal, amplify it, and then detect it. This approach is being replaced by digital sampling devices that directly sample all the electrical activity found on an antenna over very wide bands of frequencies. Soon the antenna will become the most important component of a radio monitoring station and conventional radios will become obsolete.

Quantinization

In order for an analog signal to be converted into a digital form, it must be first quantinized. This is a process of approximating a continuous range of values (or a very large set of possible discrete values) by a relatively small set of discrete symbols or integer values. This reduces the size of the information that's contained in an analog signal down to a manageable amount.

The benefit of this process can be seen in the large amount of information stored in digital formats, such as MP3 for audio or JPEG for video. Without the data compression available through quantinization, digital audio and video files would be huge. Likewise, moving un-quantinized sample data through the circuits of a DSP or software-defined radio (SDR) would place a significant amount of over-head on them. However, if the quantinization process is not precise, the resulting analog output can contain distortion errors.

In general, errors in quantinization are due to circuit noise distorting the digital signal more than any problems in the process itself. Therefore, any reduction or control of circuit noise will reduce or eliminate the problem.

Non-linearity

These are non-specific errors caused by a failure to map the input voltages to the output voltages. Like the errors found in quantinization, non-linearity errors are due to circuit noise distorting the digital signal.

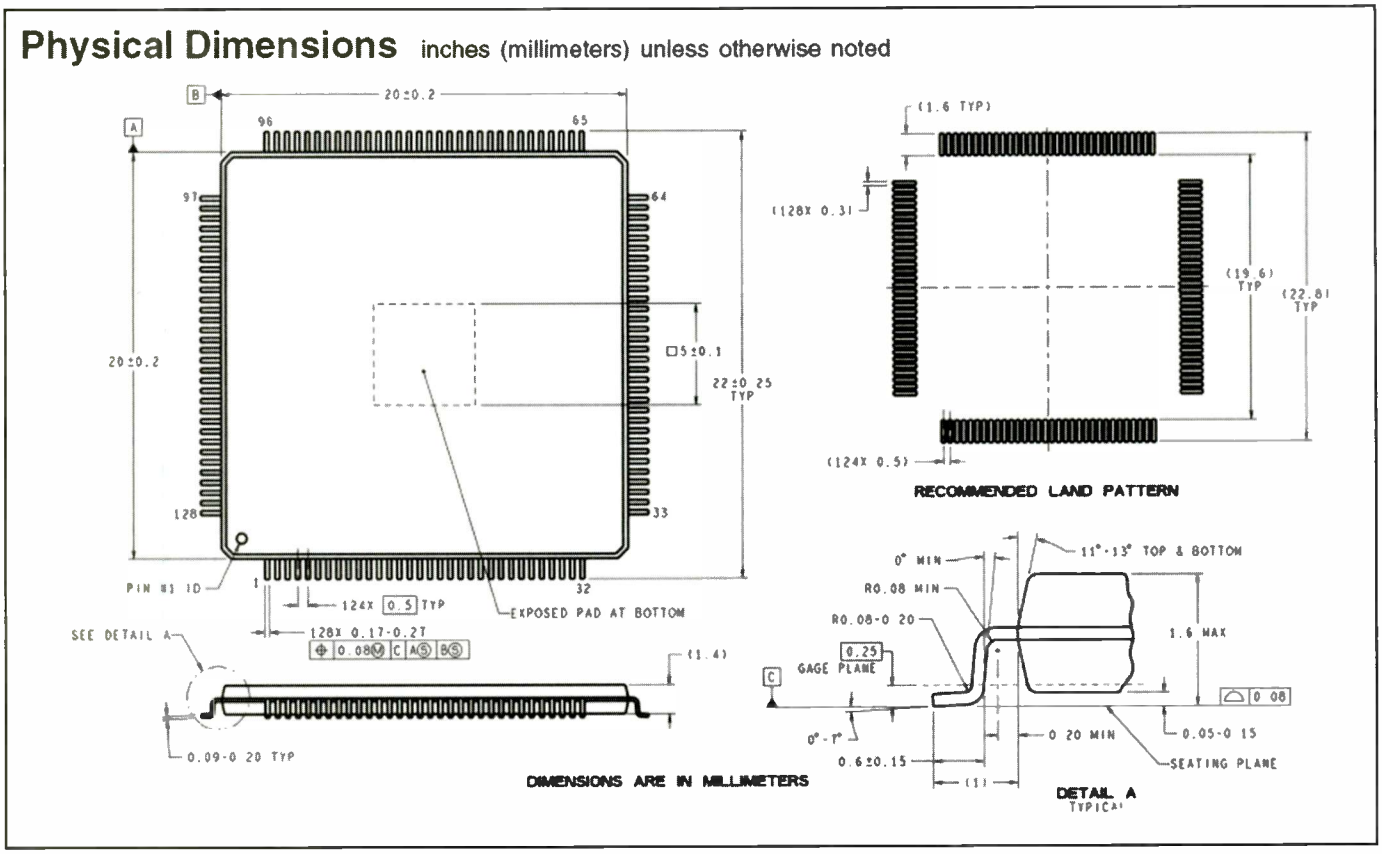
Again, hardening the digital circuit through good design and the use of quality components, or the inclusion of error correction circuitry, will generally fix the problem.

Additional Sampling Issues

The digital revolution we're now experiencing has actually taken place over a long period. It took 60 years of research and development to go from British scientist Alec Reeves outlining his theory of digital communications to the release of the Ten-Tec RX-320 software defined radio. The bottom line of the entire digital revolution has been the need for a proper technology to support it; vacuum tube and even transistor devices were simply not fast enough to sample and store large amounts of information properly.

While extraordinary projects like SIGSALY during World War II (see the June 2006 column) clearly demonstrated that digital sampling worked using vacuum tubes, it was simply too expensive to be practical. So from 1945 until the 1970s DSP and digital sampling was something that could only be created in a university laboratory by using a mainframe computer. That all changed after 1980, when increasingly sophisticated integrated circuits became available. These chips allowed for the development of the first compact disk (CD) players in 1983, which really brought digital sampling to the attention of the general public.

In the 1990s completely programmable computers on a chip became available, enabling the development of sampling devices that could sample into the radio frequency range. While limited to



Welcome to the future of radio. Here's National Semiconductor's ADC081000, the device that will soon replace conventional radios. It's an ultra-high-speed digital sampling chip that's completely self contained and capable of sampling RF frequencies between 0 to 1.6 GHz. It eliminates the need for any tuning circuits and actually provides better sensitivity than a conventional radio. It runs on less than 2 volts and uses 1.4 watts of power. There are even faster and more powerful digital sampling chips now under development. (Diagram courtesy National Semiconductor)

sampling up to 10 MHz, these chips were inexpensive and led to the development of the first DSP and software-defined radios, such as the RX-320. Today we're seeing the development of a whole new generation of digital devices that have a sampling bandwidth of up to 1.6 GHz and eliminate the need for RF tuning circuitry of any kind. You simply hook up an antenna to the device and sample whatever signal is present on the antenna.

One such device is National Semiconductor'sADC081000 chip (www.national.com/pf/AD/ADC081000.html), which comes operational on an evaluation board and costs only $1,000 and draws only 1.4 watts at 1.9 volts. While this product is not designed for the consumer market, it's definitely within the reach of the serious amateur who has the skill and technical background to use it.

Ironically, though, these sampling devices actually take radio back to the original form used by Dr. Heinrich Hertz when he discovered radio waves. His equipment consisted basically of a circular ring with a gap over which a spark was produced when radio waves passed through it. The observed phenomenon, radio waves causes the movement of electrical currents in an antenna as they pass through it, is all that's needed to receive a radio signal. Indeed, you could have a simple radio made up of nothing more than a detector and a set of headphones.

As common sense dictates, however, all you would hear over such a radio would be a jumble of signals. Remember, the only reason radio receivers have been employed has been to select a single signal out of a frequency range, amplify it, detect it and then place it into either a loudspeaker or headphones.

But today's high-speed digital sampling devices have rendered all conven-

In Memoriam—Eugene E. Collins

On June 9, 2006, Eugene Collins, KA2IWJ, of Tonawanda, New York, passed away. Gene was the president of RAWNY, the Radio Association of Western New York (W2PE), and editor of the club's newsletter *QSO*. (RAWNY was formed in 1917 and is believed to be the second oldest amateur radio group in the United States. The club station operates from *USS Little Rock* at the Buffalo and Erie County Naval & Military Park in downtown Buffalo, New York.)

Gene was extremely active in the club for many years as well as his own community. Through his generosity of spirit, he earned the respect of many people, myself included. He is survived by his wife Ruth (Kline); his sons Daniel, James, and Timothy; his brothers Philip and Stephen; and five grandchildren and several nieces and nephews.

Thanks, Gene, for inviting me to speak on radio topics at several RAWNY meetings and sponsoring me as an honorary member of the club. He will be missed.

tional radio receivers obsolete. From now on, a state-of-the-art radio monitoring station will consist of three things: a high-performance antenna, a digital sampling device connected to that antenna, and a computer program to select and detect signals within the sampled RF data. Conventional radios will be replaced by small, efficient sampling devices simply because they will no longer be able to compete with the new technology. Anyone who's used the new generation of sampling-based radios, such as those found in SDRs, has already discovered their improved selectivity, sensitivity, and audio fidelity.

These new radios also offer increasingly broad frequency ranges, access to a wide range of modes to demodulate, and integration into personal computers. The benefit of this change to the radio monitoring community is that it will open up a new age of affordable experimentation, as people will be able to build new highly efficient antenna configurations to capture fainter signals than ever before.

Rather than heralding the end of radio monitoring, the new digital sampling technology will open up new opportunities for experimentation and design. How the radio monitoring community comes to use this new technology to best advantage will depend entirely upon our willingness to learn how to use it.

That step is entirely up to you.

Coming Next Month

Next month I'll look at how sampling devices convert digital information back into analog form. This is a very critical step in the process, as any errors or problems will result in a distorted analog signal. I'll also look at more of the current digital sampling devices on the market, particularly those suitable for the amateur and consumer market.

You can e-mail me with any questions at carm_popcomm@hotmail.com. I can't answer general questions on computers, but will be more than happy to help you with any issues raised in the columns.

As I write this column it's still early summer, and we've yet to have any major storms, and there are even some indications that we may not have a repeat of last years Katrina disaster. I hope this prediction comes true, but if it doesn't please send a donation to the American Red Cross (www.redcross.org/donate/donate.html) to help your fellow Americans. There are many other organizations that you can contribute to, so please use them if you wish but do not give into "charity fatigue." As always, if you have a job, a family around you, and are living in a stable neighborhood, then frankly show your thanks for that wonderful good luck by sharing of that with someone less fortunate, and do so regularly.

Let us also not forget our troops overseas who continue to need our visible support, particularly as fighting in the Middle East and Afghanistan now seems destined to move into other parts of the world as unrest spreads. Please refer to the U.S. Department of Defense's official webpage, "Defend America." There's a section, found at www.defendamerica.mil/support_troops.html, with an amazingly wide range of practical and useful ways that you can directly help.

If you're fortunate enough to live in the United States of America, please remember to give thanks for your personal blessings by remembering to pass on that blessing to others through regular acts of selfless sharing. ■

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Tuning In *(from page 4)*

on the rails!), but I usually cringe when just walking *across* the NJ Transit tracks. If railfans are going to be an accepted part of the rail landscape, it makes sense to have designated areas set aside for railfan hobbyists—perhaps scenic spots, parks, cow pastures, bike trails, overpasses and other safe havens where their kids and grandkids can join them and learn a healthy respect for all that steel. Just not in rail yards, mingling with official track personnel, okay? We're all taught from a very early age that train tracks are dangerous. Period. Watching, taking notes and photos from a safe distance is perfectly reasonable, but scouting around private rail company property is another matter, regardless of how much hobby experience is under your belt.

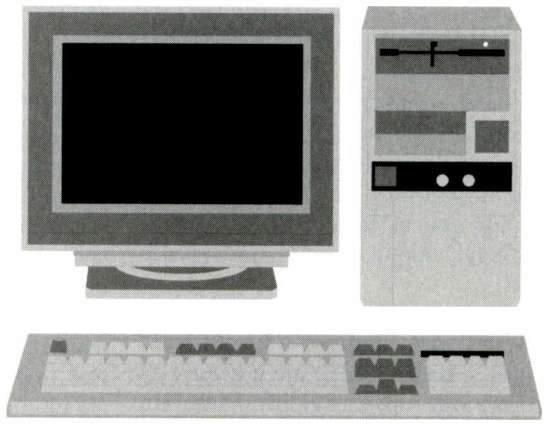
If you're a railfan, remember that the cops and rail employees are simply doing their jobs. They're just human, and it isn't their job to distinguish between the railfan with 40 years of railfanning "experience" and the doofus with a bomb in the camera bag. (Just how does one distinguish between a *terrorist* wearing a baseball cap, carrying a camera bag and scanner strapped to his belt and a railfan with the same gear? If the answer is that the ordinary railfan is cool because he's "registered" with the Citizens United For Rail Security, that's just not good enough.

Personally, deep down I'm a railfan, but there just aren't enough hours in the day for everything. And you can bet your new scanner I'm not going to be found on the tracks or in a rail yard. Nevertheless, I still remember the old black coal-burning locomotive pulling the boxcars and coal cars on the FJ&G (Fonda, Johnstown, and Gloversville) line behind the house, and the time the *engineer* came to Dad's garage and I learned *he* was the engineer I was waving to every day. (I was still too young to understand why he didn't actually bring the train down our driveway to see Dad!)

Yes, I got a Lionel train one Christmas, which is how many railfans got started in a lifelong love of trains. Mine is sitting in a box in Mom's attic. The track was a little rusted the last time I checked, and the cars are probably all cob-webbed together. Maybe I'll take some time to check out that dusty box and ask Mom to tell me again about how we'd sit for hours and wave at the trains when life was a lot simpler. I'll bring the scanner along, but there's no train close enough to hear. I'll also show Mom my new ID, but too bad it won't help me renew next year at New Jersey's DMV. ■

by Joe Cooper, carm_popcomm@hotmail.com

Digital-to-Analog—The Technical Aspect Of The Conversion Process



On the morning of December 7, 1941, American code breakers uncovered startling information in some apparently routine Japanese diplomatic traffic: Japan had apparently given up on achieving a diplomatic solution to the growing problems it was having with the United States, and war seemed to be the only logical outcome.

The news quickly reached U.S. Army Chief of Staff General George Marshall, who was then faced with a serious decision. It was obvious that the new information needed to be disseminated as quickly as possible, but the question was which method of communications to use. Voice was the obvious choice for speed, but Marshall had serious misgivings about using this method of communication.

The only secure method the United States was using at that time was an analog device maintained by Bell Labs, known as the A-3 voice scrambler, which had been developed in the 1930s. The A-3 system was relatively simple, basically mixing the information signal with a second one containing noise, making it all but impossible to understand. This noise was removed from the signal at the receiving end by mixing it with a new signal containing identical noise, subtracting (or canceling) out the original noise.

Unfortunately, one or more German engineers were involved in the A-3 project during its development and understood the theory behind the technology. Therefore, when World War II began Germany knew exactly how the scrambling technology worked (though in reality it had been far from secret in the first place). As a result, Deutsche Reichspost, the government agency responsible for all telephone and telegraph transmissions in Germany and occupied Europe, developed effective methods for breaking the A-3 system by the early 1940s. Using an interception site located on the Dutch Coast, German intelligence was able to listen in to extremely sensitive military and diplomatic communications of the Allied forces, including those taking palace between President Franklin Roosevelt and Prime Minister Winston Churchill.

Since the Japanese were allied with Germany, Marshall was certain that they too had access to the technology needed to crack the A-3 system. So rather than using voice transmission, he opted for a coded radiotelegraph—and the message arrived after the sneak attack on Pearl Harbor had begun.

However, two years later, in the summer of 1943, the same German listening post that had been monitoring A-3 signals began to hear a new signal. It sounded like the buzz of an insect. At first the intelligence officers believed it to be a new type of high-speed telegraph signal. The Germans had used high-speed telegraph technology before the World War I in the form of a magnetic wire recorder. In that system, the original telegraph messages were transcribed to the wire recorder at normal speed. The recording was then played over a transmitter at a speed many times faster than could be achieved by a human operator, thus reducing the actual time required for transmission of a large amount of information. At the receiving station, another wire recorder, operating at the same speed as the one used in the transmission, recorded the transmission. When played back at a greatly reduced speed, the message was again intelligible and could be transcribed.

Believing this was what they were hearing, German intelligence officers dutifully recorded the transmission of these new signals and attempted to use the same technique of decoding them by slowing down the speed of the recorded signal. However, no matter how much they reduced the speed, they were unable to uncover anything remotely approaching a decipherable telegraph code. Undaunted, German intelligence continued to record countless hours of this buzz; recordings which the Allied forces later discovered when they overran the Dutch listening post.

A Radio First

What the Germans didn't realize was that what they were listening to was not telegraph code, but the first digitally sampled voice transmission. Code-named SIGSALY by the Allies, this new mode of voice communication was the first completely unbreakable technology to be devised for intelligence work. It was unbreakable because the data encryption made it undecipherable, rather than simply scrambling it with noise, as was done with analog technology.

The level of encryption provided by SIGSALY was so secure that Roosevelt and Churchill were once again able to speak to each other directly over a telephone, allowing vital ideas and information to be shared quickly. Thanks to the timely introduction of digital voice technology, the Allies were able to execute complex military maneuvers without having German or Japanese intelligence intercept and decode them. That ability to encrypt voice transmissions using digital technology, along with other important scientific advances, gave the United States and her Allies the advantage they needed to win World War II.

Making Digital Intelligible

The story of SIGSALY holds an important lesson for those of us who want to understand the practical application of digital technology in radio communication: it's simply that human

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senses cannot use digital information. In fact, our senses of hearing and sight are very limited in their ability to process digital information. Anyone can witness this just by watching someone trying to master telegraphic code for a ham radio license. Of course, the human brain is very capable of deciphering extremely complex codes and patterns, but the time it takes to do this is can be very long. You can see this in watching chess players during a game, or reading about the solving of a mathematical riddle that took the best mathematical minds decades to unravel.

Yet today even the most inexpensive home computer can fly through, in less than a second, mathematical calculations that would have taken a human being months to work out. That, or course, is because today's computers can process millions of bits of information each second, something even mainframe computers had difficulty accomplishing 30 or 40 years ago. This cheap and accessible computer processing power now gives us easy access to digital signal processing (DSP) technology.

DSP is a critical component in any method of turning analog sound or video into digital information. When any digital mode is applied to analog voices, images, or data, the resulting flow of data is heard as that buzzing sound that engineers working on the SIGSALY project had dubbed "the Green Hornet," after the famous 1930s radio show of the same name.

The Underlying Difference

Let's look at the technical issues involved in converting a digital signal back into an analog form that human senses can understand. This is not a simple process, as German Intelligence quickly discovered, as there's no ready clue found in a digital data stream to help you decode the intelligence found in there.

So exactly how are the zeros and ones of digitally sampled information transformed back into analog form. Complicating the process is that there can be a wide range of problems encountered that can cause distortion and noise. Anyone who's worked with digital technology knows that the mode is not 100-percent noise free. A surprising number of clicks, crackles, skips and other noises can be introduced when a digital recording is made. The same is true for the reverse process, where digital signals are returned to their analog form.

Regardless of the technology employed—analog or digital—the goal for any method of communication is the same as that articulated by scientist Claud Shannon in his paper, *A Mathematical Theory of Communication*, published in the late 1940s. Shannon stated, "The fundamental problem of communication is that of reproducing at one point either exactly or approximately a message selected at another point."

What makes digital technology superior is that it can deliver an "exact or approximate" reproduction better than analog can, even with its potential for noise and distortion problems. Digital transmission is the preferred choice among communications specialists because of its primary characteristics, which are as follows:

- The quality of the received signal is equal to the state of the original signal.
- The quality of the received signal is independent of transmission media.
- The method of transmission is completely compatible with different media and intelligence (video, audio, and data).
- The technology involved in digital transmission and reception is low cost.
- New features can be embedded into digital transmissions very easily.

The bottom line is that analog transmission techniques cannot deliver the same results as digital. With analog technology, the quality of the received signal is not the same as the original unless very expensive technology is employed to remove the noise that is always present in its transmission and reception.

Since noise is generally an *occasional* event with digital technology, rather than a constant characteristic as it is in analog, let's see how one keeps those noise events to a minimum. More importantly, let's also see how digital samples are successfully converted back into their original analog form and how they can actually sound and look better than the original, thanks to the employment of DSP technology.

Basics Of DSP

Two devices are used to convert digital signals back to their analog form: a Codec (for compressor-decompressor), also referred to as a Coder-Decoder or compression/decompression algorithm, which is DSP software; and a hardware-based digital-to-analog converter (DAC). The primary function of a DAC is to change numbers (digital information) into a voltage that corresponds to the value of those numbers. A DAC only works with pulse code modulation (PCM) encoded signals.

PCM was invented by the British engineer Alec Reeves in 1937 while working for the International Telephone and Telegraph in France. Later, Bell Labs would discover PCM independently from Reeves when they created the SIGSALY device, although Reeves received the U.S. patent for the technology in 1943. Interestingly, recent historic research has uncovered evidence that digital sampling techniques using mechanical (rather than electronic) devices were employed as early as 1853 (I'll touch on that in a future column).

Theoretically PCM is a very simple technique, wherein a signal is sampled (that is, has its voltage level measured) at a precise interval and that sample is then converted into a binary value made up of 0s and 1s. While that sounds simple enough, it's relatively tricky to create a perfect sample because errors can find their way into the calculation of the digital values and because of the frequency range of the sample itself. If any errors do take place in the sampling, the result is very distinct noise (pops and clicks) or distortion in the resulting digital signal (see **Figure 1**).

Fortunately there are several techniques available to minimize sampling errors. But generating a properly sampled PCM data stream (sometimes referred to as "essence") is only half the job; there are just as many potential problems when it comes to converting digital information back into an analog form. It's when the PCM data is in its sampled form, however, that the real value of digital technology becomes apparent.

Remember that Shannon postulated that ideal communication takes place when you have "exact or approximate" reproduction at the received end. The key word here is "approximate"; you do need a perfect representation of the original information (sound or data) to have achieved successful communication. So practically all digital sampling techniques aim to *approximate* the analog signal through data processing, which is where the Codec comes into play.

The Codec processes the digital information to reduce as much as possible the number of 0s and 1s of the original sampling. When the total amount of digital information is

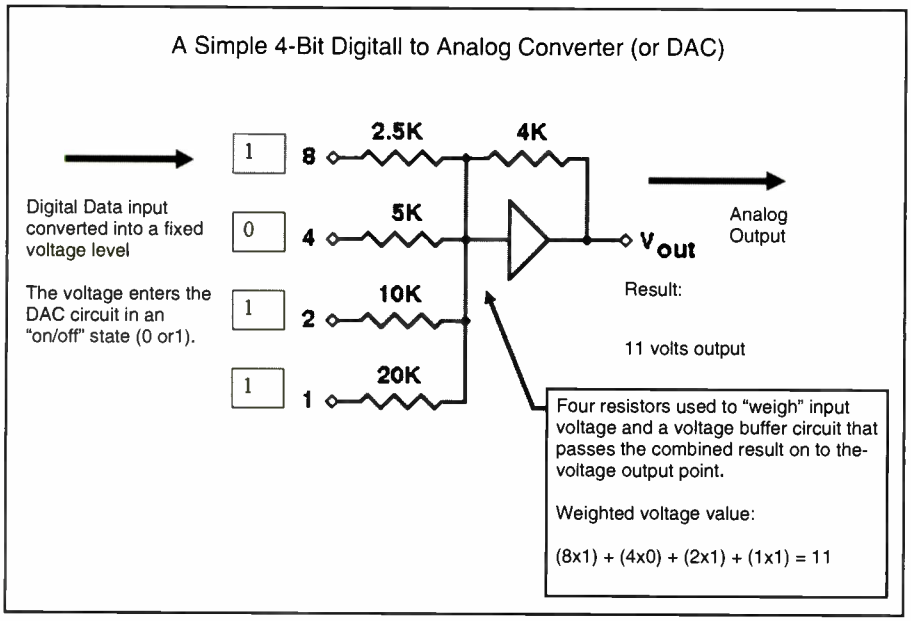

Figure 1. An illustration of two possible errors that can occur in a DAC when digital and analog processing is not linear.

reduced, the chances for errors occurring in that data are reduced significantly. Codecs do this through different data compression techniques, with specific techniques used for specific results. There are actually thousands of types of Codecs that can be used in DSP. For example, a Codec could be chosen for audio sampling that emphasized bass sounds over treble. Or in the case of video, a Codec could be used to produce good motion reproduction (for a sports event) over color. Another Codec could be used to emphasize color over motion (for a drama with close-ups of the actors' faces).

While data compression of a Codec does reduce errors, it also reduces the quality of the signal because of the reduction of information contained in it (digital systems are inherently restricted in the amount of bandwidth available for DSP). The sad reality, then, of DSP is that the Codec used for data compression will always result in a final output that is compromised in some way.

Fortunately, the average human ear, or eye, or brain is not sensitive enough to actually perceive that reduced quality so such data losses are acceptable. In fact a well-designed Codec can introduce "wrapper" data into the data stream that can overcome those losses, creating a new stream of "metadata" that can look or sound even better than the original.

So when digital information is ready for conversion back into analog form, it's no longer in its original sampled "essence," nor is it PCM. It is now a "metadata" signal. For the DAC to use that signal, it must first be decompressed by passing it back through the Codec, which decodes it back into PCM. Once that has been done, the DAC can then turn it into an analog signal.

How The DAC Operates

Ideally, a DAC operates in a purely linear manner, so that whatever value is presented to it in digital form is accurately represented in its analog output. To achieve that linearity, the DAC must process the digital information into analog form at precisely the same clock rate at which the original analog information was converted into digital information. Of course, no matter how well a DAC is designed, it will never be able to achieve an idealized linear output.

Remember, the design goal for a DAC is to achieve an "approximate" representation, which is often simply a matter of making the best choice of the DAC methodology for a given application. The basic design of a DAC is actually very simple. **Figure 2** shows a 4-bit DAC designed to convert digital information into a voltage. On the left of the diagram is a series of voltage inputs that are determined by the digital information presented. The range of the digital information is from 0000 to 1111, which when "weighed" by the DAC will produced voltages ranging from 0 to 15 volts, or a total of 16 possible voltage levels (4 x 4).

How the voltage values are actually produced is based upon the weighted value of each of the digital place holders in the circuit. In this particular case the weighted values are 8, 4, 2, and 1, where each of those values is multiplied by either 0 or 1. So if a digital value of 1111 is presented, then the total weighted value would be (8x1)+(4x1)+(2x1)+(1x1) = 15 volts output. Likewise, if the value presented is 1011, then the total weighted value would be (8x1)+(4x0)+(2x1)+(1x1) = 11 volts output.

The actual weight in the circuit is determined by four resistors, each providing the necessary voltage drop for the weight they represent in the equation. If you refer to **Figure 3** you will see that as the digital values change, the voltage output changes, and it does so in a linear manner.

Of course in the real world, DAC circuits are much more complicated than this, and there are a great many different types an engineer can choose from. You'll find descriptions of over sampling DACs, Binary Weighted DACs, R-2R Ladder DACs, and even Hybrid DACs when reviewing the literature on the subject. Nevertheless, the basic principles I've outlined apply in even the most complex examples. There is always a digital value being converted into an analog signal (generally a voltage) in as linear a fashion as possible, and with as few errors and distortions as can be achieved within a particular design.

DAC Characteristics

So now that we have our basic DAC circuit, how do you measure or describe its operation? There are many ways to do so, but there are five that are most commonly used:

Resolution—The range of information that can be represented, generally stated as the number of bits it uses. In our example circuit, the resolution is 2 to the 4th power, or 16. If there were 8 bits, it would be 2 to the 8th power, or 256. Chips exist today that have a resolution greater than 17 bits, which yield than 262,144 levels of resolution.

Maximum Sampling Frequency—As stated in the Shannon-Nyquist sampling theorem, a signal must be sampled at over twice the bandwidth of the desired signal. So if you're sampling in the average audio range of the human ear (20 kHz), the sampling frequency would have to be over 40 kHz. The standard used for audio CDs is 44.1 kHz, so the maximum sampling frequency for most audio DACs is that value.

Monotonicity: This refers to the abil-

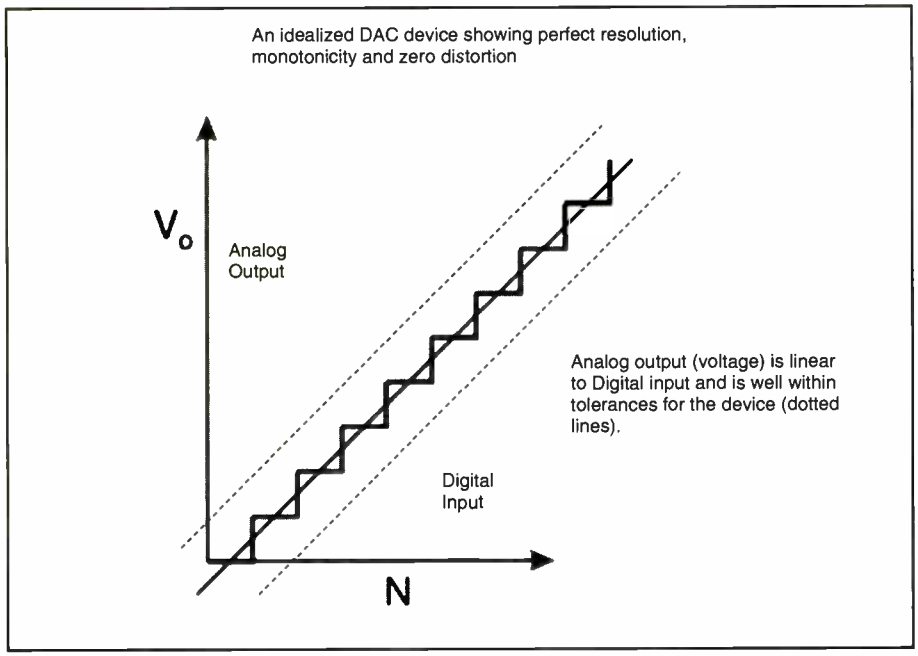
An idealized DAC device showing perfect resolution, monotonicity and zero distortion
Vo
Analog Output
Analog output (voltage) is linear to Digital input and is well within tolerances for the device (dotted lines).
Digital Input
N

Figure 2. An illustration of the ideal linear relationship between digital and analog signals that would be found in a "perfectly designed" DAC.

ity of a DAC's analog output to increase with an increase in digital code, or the converse. The term comes from the study of order between sets of numbers in calculus. Good monotonicity is achieved when digital and analog functions are properly balanced.

THD+N: This formula represents the percentage of distortion and noise the DAC introduces to the signal. This is a very important DAC characteristic. Obviously the lower the percentage the better.

Dynamic Range: This is a standard measurement employed in many electronic amplifying devices. It is the difference between the largest and smallest signals the DAC can reproduce. It's expressed in decibels.

After the DAC has converted the digital signal to analog form, all that is generally required is a conventional amplifier and filter to bring the resulting signal up to usable form (say, to drive a speaker or headphones, if the signal is audio).

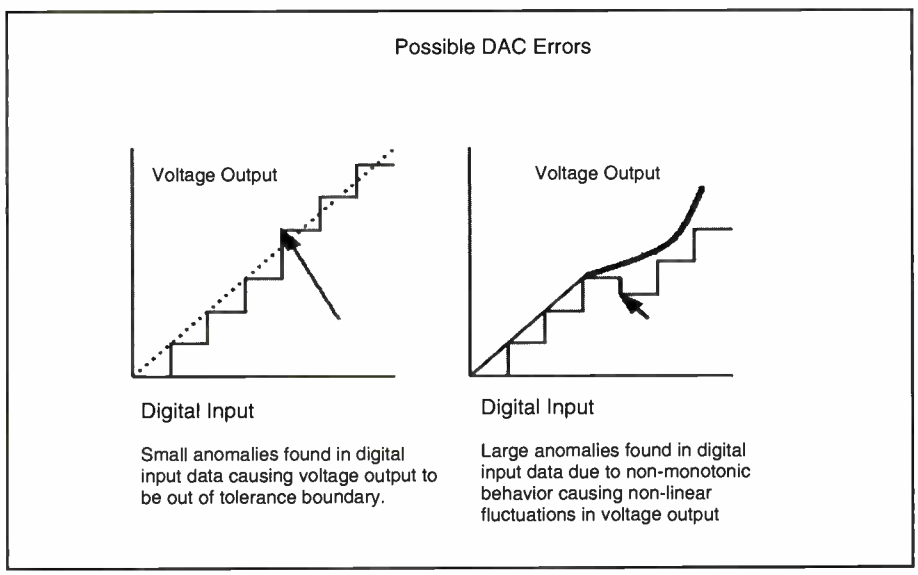
Possible DAC Errors
Voltage Output
Digital Input
Small anomalies found in digital input data causing voltage output to be out of tolerance boundary.
Voltage Output
Digital Input
Large anomalies found in digital input data due to non-monotonic behavior causing non-linear fluctuations in voltage output

Figure 3. This simple circuit illustrates the basic function of a DAC, as outlined in this month's column.

The simplest and most prevalent DAC in commercial use is probably the sound card in your computer. So if you want to know more about this subject, simply learn more about that device, which also happens to be an ADC (analog-to-digital converter). In future columns I'll be looking at computer sound cards in more detail as they're actually very good laboratories for learning about DSP and other data processing functions. We'll look specifically at how to use the Microsoft Visual Basic development environment to create your own DSP software programs to use with your sound card.

Coming Next Month

Next month I'll return to computer programming and look again at how to use Microsoft's Visual Basic to control Ten-Tec's RX-320/320D.

For now, you can e-mail me directly with any questions at carm_popcomm @hotmail. com. I can't answer general questions on computers, but will be more than happy to help you with any issues raised in the columns.

As I write this, we have still not been hit by a major storm, but if that changes, please send a donation to the American Red Cross at www.redcross.org/donate/donate.htm. Of course, there are other good (and ethical) organizations you can contribute to if you don't wish to use the Red Cross, so please use them if you wish, but try not give into "charity fatigue." As I've said many times before, if you do have a job, a family around you, and are living in a stable neighborhood, then frankly show your thanks for that wonderful good luck by sharing with someone less fortunate, and do so regularly.

Let's also not forget our troops overseas, especially during the holiday season. They continue to need our visible support, particularly as fighting in the Middle East and Afghanistan now seems destined to move into other parts of the world. Please refer to the U.S. Department of Defense's official webpage, "Defend America." The section found at www.defendamerica.mil/support_troops.htm has an amazingly wide range of practical and useful ways that you can directly help.

Again, if you are fortunate enough to live in the United States of America, please remember to give thanks for your personal blessings by passing on that blessing to others through regular acts of selfless sharing.

See you again next month! ■